Lucent Technologies
Bell Labs Innovations

DEFINITY®

Enterprise Communications Server

Release 8.2
Overview

555-233-002
Issue 1
April 2000

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Notice

Every effort was made to ensure that the information in this book was complete and accurate at the time of printing. However, information is subject to change.

Your Responsibility for Your System's Security

Toll fraud is the unauthorized use of your telecommunications system by an unauthorized party, for example, persons other than your company's employees, agents, subcontractors, or persons working on your company's behalf. Note that there may be a risk of toll fraud associated with your telecommunications system and, if toll fraud occurs, it can result in substantial additional charges for your telecommunications services.

You and your system manager are responsible for the security of your system, such as programming and configuring your equipment to prevent unauthorized use. The system manager is also responsible for reading all installation, instruction, and system administration documents provided with this product in order to fully understand the features that can introduce risk of toll fraud and the steps that can be taken to reduce that risk. Lucent Technologies does not warrant that this product is immune from or will prevent unauthorized use of common-carrier telecommunication services or facilities accessed through or connected to it. Lucent Technologies will not be responsible for any charges that result from such unauthorized use.

Lucent Technologies Fraud Intervention

If you *suspect that you are being victimized* by toll fraud and you need technical support or assistance, call Technical Service Center Toll Fraud Intervention Hotline at 1 800 643-2353 or contact your Lucent Representative.

Federal Communications Commission Statement

Part 68: Statement

Part 68: Answer-Supervision Signaling. Allowing this equipment to be operated in a manner that does not provide proper answer-supervision signaling is in violation of Part 68 rules. This equipment returns answer-supervision signals to the public switched network when:

- Answered by the called station
- Answered by the attendant
- Routed to a recorded announcement that can be administered by the CPE user

This equipment returns answer-supervision signals on all DID calls forwarded back to the public switched telephone network. Permissible exceptions are:

- A call is unanswered
- A busy tone is received
- A reorder tone is received

Lucent Technologies attests that this registered equipment is capable of providing users access to interstate providers of operator services through the use of access codes. Modification of this equipment by call aggregators to block access dialing codes is a violation of the Telephone Operator Consumers Act of 1990.

This equipment complies with Part 68 of the FCC Rules. On the rear of this equipment is a label that contains, among other information, the FCC registration number and ringer equivalence number (REN) for this equipment. If requested, this information must be provided to the telephone company.

The REN is used to determine the quantity of devices which may be connected to the telephone line. Excessive RENs on the telephone line may result in devices not ringing in response to an incoming call. In most, but not all areas, the sum of RENs should not exceed 5.0. To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company.

NOTE:

REN is not required for some types of analog or digital facilities.

Means of Connection

Connection of this equipment to the telephone network is shown in the following table.

Table 1. Means of Connection

Manufacturer's Port Identifier	FIC Code	SOC/REN/A. S. Code	Network Jacks
Off/On Premises Station	OL13C	9.0F	RJ2GX, RJ21X, RJ11C
DID trunk	02RV2-T	0.0B	RJ2GX, RJ21X
CO trunk	02GS2	0.3A	RJ21X
CO trunk	02LS2	3.0A	RJ21X
Tie trunk	TL31M	9.0F	RJ2GX
1.544 digital Interface	04DU9-B,C	6.0F	RJ48C, RJ48M
1.544 digital Interface	04DU9-BN,KN	6.0F	RJ48C, RJ48M
2.048 digital Interface	04DU9-BN,KN	6.0F	RJ48C, RJ48M
120A2 channel service unit	04DU9-DN	6.0F	RJ48C

If the terminal equipment (DEFINITY® System) causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice is not practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.

If trouble is experienced with this equipment, for repair or warranty information, please contact the Technical Service Center at 1-800-242-2121. If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.

It is recommended that repairs be performed by Lucent Technologies certified technicians.

The equipment cannot be used on public coin phone service provided by the telephone company. Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission or corporation commission for information.

This equipment, if it uses a telephone receiver, is hearing aid compatible.

Canadian Department of Communications (DOC) Interference Information

This digital apparatus does not exceed the Class A limits for radio noise emissions set out in the radio interference regulations of the Canadian Department of Communications.

Le Présent Appareil Nomérique n'émet pas de bruits radioélectriques dépassant les limites applicables aux appareils numériques de la class A préscrites dans le reglement sur le brouillage radioélectrique édicté par le ministére des Communications du Canada.

European Union Declaration of Conformity

The "CE" mark affixed to the DEFINITY® equipment described in this book indicates that the equipment conforms to the following European Union (EU) Directives:

- Electromagnetic Compatibility (89/336/EEC)
- Low Voltage (73/23/EEC)
- Telecommunications Terminal Equipment (TTE) i-CTR3 BRI and i-CTR4 PRI

For more information on standards compliance, contact your local distributor.

Table 1. ETS standards referenced by iCTR3 and CTR4

	iCTR3	iCTR4	CTR3	CTR4
L1:	ETS300012	ETS300011	ETS300012	ETS300011
L2:	ETS300153	ETS300156	ETS300125	ETS300125
L3:	ETS300104	ETS300104	ETS300102	ETS300102
Safety:	ETS300047	ETS300046	ETS300047	ETS300046

How to get help

If you need additional help, the following services are available. You may need to purchase an extended service agreement to use some of these services. Contact your Lucent representative for more information.

DEFINITY Helpline +1 800 225 7585
(for help with feature administration and system applications)

Lucent Technologies National Customer Care Center Support Line +1 800 242 121
(for help with maintenance and repair)

Lucent Technologies Toll Fraud Intervention +1 800 643 2353

Lucent Technologies Corporate Security +1 800 822 9009

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About This Book

What Is the Purpose of This Book?

This book provides general information about the components and capabilities of the DEFINITY® Enterprise Communications Server (referred to as DEFINITY ECS or the system). It also discusses practical and creative applications for the DEFINITY ECS platform.

This document covers information related to DEFINITY ECS R8.2. For details about changes for Release R8.2, refer to *DEFINITY Enterprise Communications Server Release R8.2 Change Description, 555-233-411, Issue 1*.

Who Should Read This Book?

This book is written for those who are considering the purchase of a DEFINITY ECS system and for Lucent Technologies representatives and distributors who need high-level information about the system and how it can be used.

What Is in This Book?

This book discusses all DEFINITY capabilities available world-wide. It defines common, practical solutions and suggests unusual, creative ones.

NOTE:
Some products are unavailable in some countries. Please check with your local distributor for further information about which features and solutions are available to you.

Conventions Used in This Book

The following conventions are used in this book:

- The word "system" is a general term for the DEFINITY Enterprise Communications Server.
- The information in this book refers to DEFINITY ECS R8 unless otherwise specified.

Trademarks and Service Marks

This book contains references to the following Lucent Technologies trademarked products:

- AUDIX®
- Callmaster®
- CallVisor®
- CenterVu™
- CONVERSANT®
- DEFINITY®
- INTUITY™
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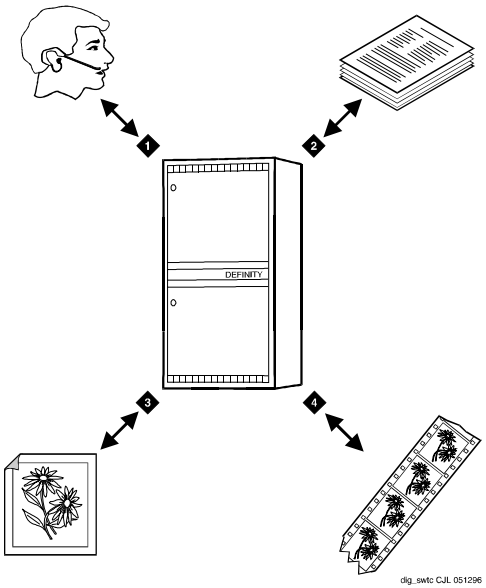
1 —System Overview

DEFINITY Enterprise Communications Server (ECS) organizes and routes voice, data, image and video transmissions. It can connect to private and public telephone networks, Ethernet LANs, ATM networks, and the Internet.



1 Voice
2 Data
3 Image
4 Multimedia

Figure 1. DEFINITY Enterprise Communications Server

Hardware

Though the primary components are the same, your DEFINITY ECS can vary widely in size and appearance, depending on your capacity requirements. It may be as small as a single wall-mounted cabinet, or it may be as large as several tall cabinets linked together in the same room or even hundreds of kilometers apart. Regardless of configuration, however, the system's footprint is relatively small.

The DEFINITY system is essentially an arrangement of port networks and circuits that connect incoming to outgoing communications ports. Up to three port networks can be connected directly to each other. When there are more than three port networks, the connections are made through a Center Stage Switch.

Processor Port Network (PPN)

Every DEFINITY ECS has one Processor Port Network (PPN). It is often the only component in small systems. The PPN houses the Switch Processing Element.

The Switch Processing Element (SPE) contains the central processing unit, which supervises system operation. It also contains a mass storage system for loading system software and saving system translations.

Because your application requirements may vary widely, DEFINITY ECS has three types of SPEs available with proven capacities of 70,000 calls per hour, 140,000 calls per hour, and 250,000 calls per hour. The performance you realize will depend on the call processing, administrative, and maintenance activities in which your system is engaged.

Expansion Port Network (EPN)

Expansion Port Networks (EPNs) are used when the system grows beyond the capacity of a single port network or must serve geographically dispersed offices. EPNs provide additional ports as needed. A system can have up to 43 Expansion Port Networks.

Center Stage Switch (CSS)

The Center Stage Switch (CSS) is a connection hub that provides port network communication. It is an essential component of a DEFINITY ECS configuration if the system is composed of more than three port networks. Often it is incorporated into smaller configurations to allow for growth. The CSS consists of from one to three switch nodes. Switch nodes are composed of one or two switch node carriers, depending on

whether the system is being duplicated for enhanced reliability. Each carrier can reside in the PPN cabinet or an EPN cabinet. One switch node can accommodate up to 15 Expansion Port Networks.

Fiber Link Administration

Port cabinets are connected via direct fiber links or through fiber links to a Center Stage Switch to provide the connections required for voice and data information transfer. The CSS is composed of switch node carriers that are interconnected by fiber links. It provides both circuit-switched and packet-switched connections. Fiber Link Administration creates the translation data defining these links by identifying the endpoint pairs for each link. Endpoints can be an expansion interface or a switch-node-interface circuit pack.

Carriers and Cabinets

Carriers are enclosed shelves composed of vertical slots that hold circuit packs. Circuit packs make up the logic, memory, and switching circuitry for the system. Port circuit packs connect to telephones, computers, and communications lines. The carriers are designed to accept any type of port circuit pack in each circuit pack position.

Each cabinet contains at least one carrier. The circuit packs fit into connectors attached to the rear of the slots. Every connector is connected to signal buses and power supplies in the cabinet.

The cabinets also house equipment that supplies power backup, ringing signal voltage, and mass storage for software translations.

There are three types of cabinets

- Compact Modular Cabinet. (CMC)
- Single-Carrier Cabinet. (SCC)
- Multi-Carrier Cabinet (MCC))

Compact Modular Cabinets

The Compact Modular Cabinet (CMC) is a small, wall- or floor-mounted unit that supports a single port network. It is thus well-suited to small, growing organizations. For a modest initial investment, the CMC gives you the capacity that you need now. As your organization grows, you can connect up to two more modular cabinets to expand the existing port network. Each CMC can hold up to 10 circuit packs. Figure 2 shows a Compact Modular Cabinet.

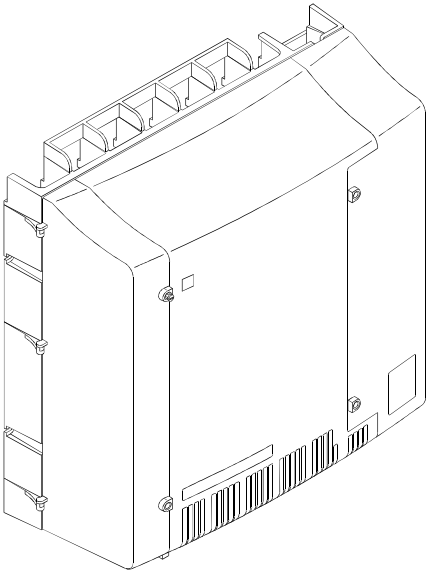


Figure 2. Compact Modular Cabinet

Single-Carrier Cabinets

The Single-Carrier Cabinet (SCC) is a compact, floor-mounted modular unit for businesses that will at some point require more capacity than Compact Modular Cabinets can supply. The SCC can support additional, expansion port networks, so you can add capacity as needed, up to the limit set by the overall DEFINITY system architecture. Each SCC contains a single circuit pack carrier with up to 20 circuit packs, depending on the carrier type. Up to four SCCs can be stacked to form a single port network, and additional port networks can be connected for still more capacity. Figure 3 shows a typical single-carrier cabinet.

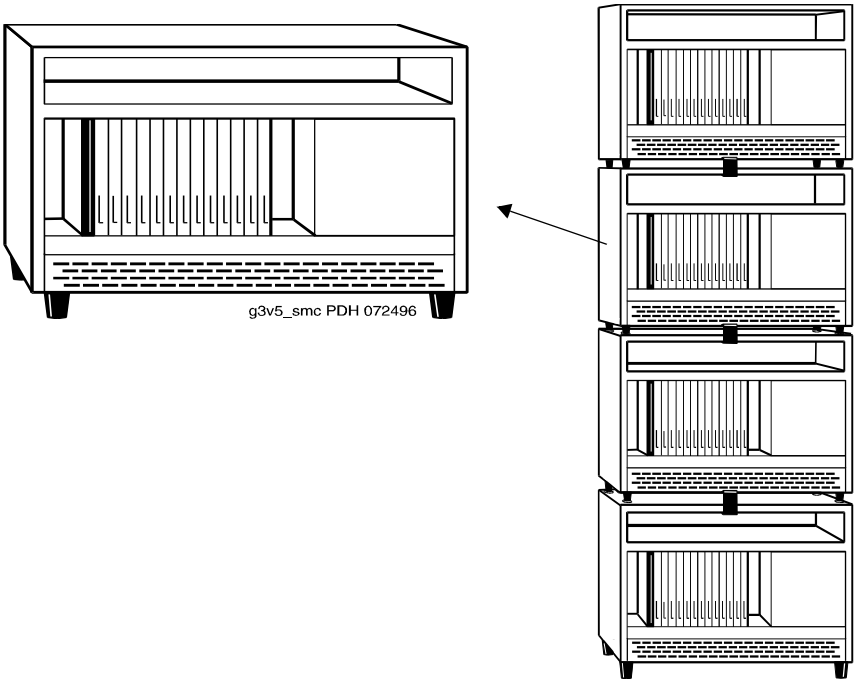


Figure 3. Typical Single-Carrier Cabinet and cabinet stack

Multi-Carrier Cabinets

The Multi-Carrier Cabinet (MCC) is a large unit designed for high-capacity communications systems and large businesses. It can contain up to five carriers and can be connected to additional, expansion port networks. Figure 4 shows a typical multi-carrier cabinet.

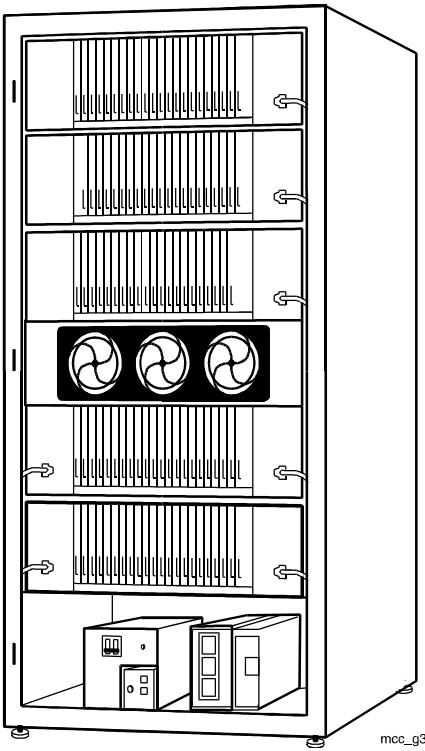
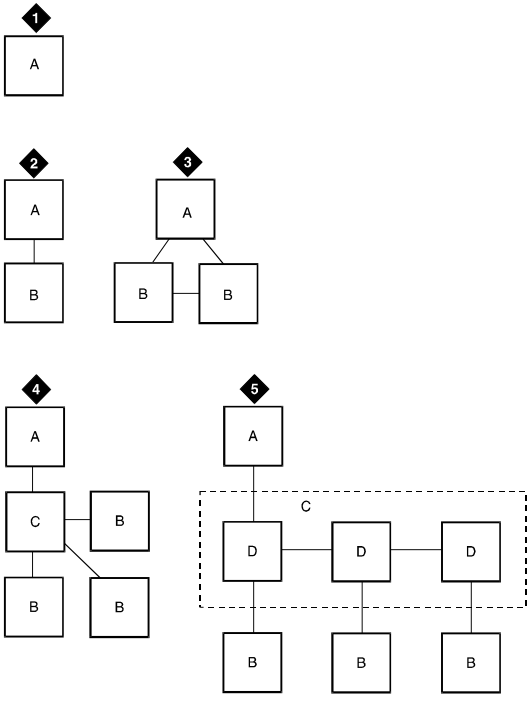


Figure 4. Typical Multi-Carrier Cabinet



1	Basic System	A	Processor Port Network
2	Directly Connected System	B	Expansion Port Network
3	Directly Connected System with Two EPNs	C	Center Stage Switch
4	CSS-Connected System with up to 15 EPNs	D	Switch Node
5	CSS-Connected System with up to 43 EPNs		

Figure 5. Standard Configurations

Reliability and Recoverability

The system is designed to recover from a power outage or other failure instantly, regardless of the source of the failure. Each port network includes a set of segmented, parallel buses. If one of the paired segments fails, the other bus segment continues to handle communications. You can always further enhance the system's reliability by duplicating critical components such as processors or fiber-optic links between port networks.

Configuring for Reliability/Recoverability

DEFINITY ECS can be configured to meet the disaster recovery needs of any business. For example:

- Calls can be routed through an alternate DEFINITY ECS if one site is destroyed or disabled by natural or man-made disaster.
- Multimedia (voice, video, data) connections to the network can all be made redundant, in case of network failures. The system can be routed through multiple public exchanges to protect against network failures (a cable or fiber being cut, for example).
- DEFINITY's universal hardware and flexible software allow systems to be re configured quickly in emergency situations. Port networks can be added and network routing can be changed in a matter of minutes.
- The TN2211 Optical Drive will replace the present TN1656 Tape Drive. The new Optical Drive will provide the same functionality as the Tape drive, such as, storage for software upgrades, translation backups, announcement file backups, core dumps, etc. It is also faster than the Tape Drive. A full backup to the Optical Drive will take approximately 20 minutes. The present Tape Drive backup takes about 95 minutes.

Survivable Remote EPN (SREPN)

The Survivable Remote Expansion Port Network (SREPN) allows a DEFINITY ECS (R6r or later) EPN to provide service to the customer when the link to the main processor fails or is severed or when the processor or CSS fails. When the links to the DEFINITY ECS

are restored and stable, the logic switch is manually reset and the EPN is reconnected to the links from the switch. There are both command and manual resets. The resets can be done remotely at the SAT or manually at the equipment.

The SREPN must be administered separately (not as a duplicated PPN) to function in a disaster recovery scenario. It will not function as a survivable remote EPN without the administration (stations, trunks, features) to support its operation.

DEFINITY ECS Network Connections

The system can be connected to public and private networks in a variety of ways.

Public Telephone Network

DEFINITY ECS supports a full range of digital and analog connections to a Public Telephone-Network Central Office, including ISDN-PRI and ISDN-BRI.

Private Telephone Networks

DEFINITY ECS supports the many kinds of private-network connections, including:

- Digital telephone handsets
- Data-communications equipment, including data modules and data terminal equipment.
- Attendant and voice messaging systems
- Multimedia and video conferencing equipment
- Call Center systems
- Hospitality systems
- Auxiliary loudspeaker-paging and music-on-hold equipment
- Peripherals, such as journal printers and call detail-recording equipment.
- Asynchronous Transfer Mode or ATM
- IP solutions, such as IP trunking and IP softphones
- Other private-network switches

TCP/IP Networks

DEFINITY ECS can also connect to Ethernet Local Area Networks (LANs) and the Internet.

- The optional DEFINITY LAN Gateway feature lets the DEFINITY system exchange messages with LAN communications software.
- DEFINITY ECS now supports Internet Protocol (IP) trunks using the TN802B circuit pack and C-LAN card (TN799B). However, to connect to the network and do signaling over IP, only the TN799B is required.

  IP trunking is a good choice for basic, corporate voice and fax communications, where cost is a major concern. IP-trunk calls travel over a company's intranet rather than the Public Telephone Network. So, for the most common types of internal, corporate communications, IP trunks offer considerable savings.

  IP trunking is usually not a good choice for applications where calls have to be routed to multiple destinations (as in most conferencing applications) or to a voice messaging system. IP-trunk calls are compressed to save network bandwidth. Repeated compression and decompression results in a loss of data at each stage and degrades the final quality of the signal. This is not a problem in normal, corporate voice or fax calls. They go through two or three compression cycles at most. But multipoint conference calls and most voice messaging systems add too many compression cycles for acceptable quality.
- DEFINITY ECS now includes the IP Solutions feature, which supports audio and voice transmissions over a LAN or WAN. For more information about IP Solutions, see "IP Solutions" on page 56.

ATM Networks

Three kinds of Asynchronous Transfer Mode (ATM) connectivity are available with DEFINITY ECS. Any standards-compliant ATM switch can serve as the switch node connecting DEFINITY ECS port networks. In this type of configuration, the ATM switch replaces the Center-Stage Switch.

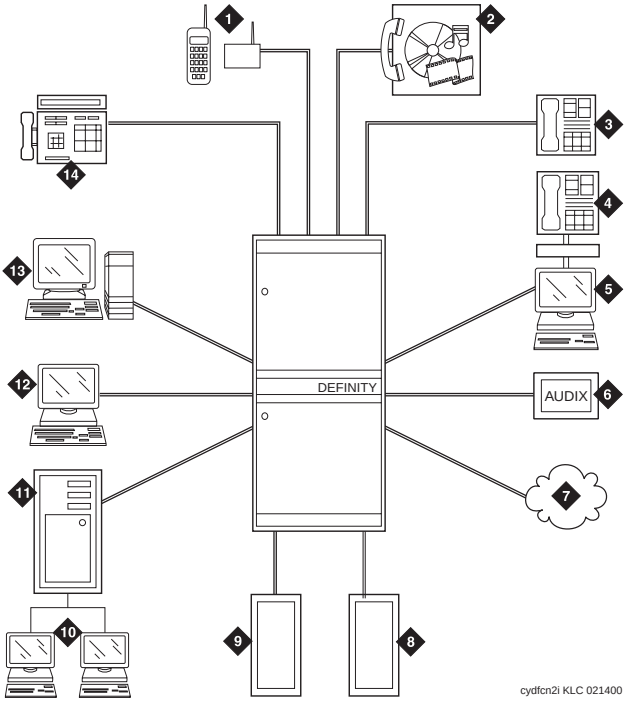
The three kinds of ATM connectivity DEFINITY ECS supports are as follows:

- ATM Port Network Connectivity (ATM-PNC) provides an alternative to either the direct-connect or Center Stage Switch configurations for connecting the Processor Port Network (PPN) to one or more Expansion Port Networks (EPNs). ATM-PNC provides reduced infrastructure cost and improved network manageability.

- ATM Circuit Emulation Service (ATM-CES) lets a DEFINITY ECS emulate an ISDN-PRI trunk on an ATM facility. ATM-CES maximizes port network capacities by consolidating trunking.
- The ATM Trunk feature supports telephony and Wide-Area Network (WAN) connectivity over ATM networks. ATM-WAN extends the PNC beyond a single ATM switch. It allows for use of DEFINITY ECS applications over an ATM infrastructure across locations. ATM-WAN consolidates communications traffic, allowing for quicker and more efficient transport of voice, data, video, and mixed media communications.

Typical Connections

Figure 6 shows typical DEFINITY ECS connections.



1	Wireless System	8	Digital Facilities
2	Multimedia Call Center	9	Analog Facilities
3	Business Telephone	10	Data Terminals
4	Telephone with Data Module	11	Host Computer
5	Data Terminal	12	Data Terminal
6	Voice Messaging System	13	Management Terminal
7	Customer LAN/WAN	14	Attendant Console

Figure 6. Typical DEFINITY ECS Connections

NOTE:
Actual equipment may appear different than the equipment shown.

International Capabilities

DEFINITY ECS provides features that allow for differences in telecommunication standards around the world, allowing you to use the same communications system at your various locations in other countries. If you are reading this book, it is likely that the system has been type approved in your country. Check with your local distributor for more information.

Public Network Call Priority

Provides call retention, forced disconnect, intrusion, mode-of-release control, and re-ring to switches on public networks. Different countries frequently refer to these capabilities by different names.

World Class Tone Detection

Enables the DEFINITY ECS to identify and handle different types of call progress tones, depending on the system administration. You can use the tone detector and identification to display on Data Terminal Dialing and to decide when to send digits on trunk calls through Abbreviated Dialing, ARS, AAR, and Data Terminal Dialing.

World Class Tone Generation

Allows you to define call-progress tones. You can select values for frequency and cadence. If you do not define a call-progress tone, DEFINITY ECS sends a default sound for that call-progress tone that is based on your administered country code. For a very few tones, that default may be silence, but for most commonly used tones, that default will make a sound.

Administrable Loss Plan

Provides you with the capability to administer the loss or gain applied on calls. This capability is necessary because some country regulations change about how much loss is allowed on voice calls.

Power

DEFINITY ECS can accept a variety of AC or DC power. The system can operate without requiring a power transformer in almost any part of the world.

During a power outage, individual cabinets (Single or Multi-Carrier) will continue to function for up to 15 seconds; the Multi-Carrier Cabinet will function for up to 10 minutes without power, depending on configuration. Optionally, an Uninterruptible Power Supply can protect a DEFINITY ECS system from under or over-voltage conditions, line frequency fluctuations, and power blackout of short duration. A battery backup system can be used to provide power for up to 8 hours, depending on the type and quantity of circuit packs and amount of traffic during the holdover period.

Software

All DEFINITY ECS systems throughout the world use the same basic software. In addition to the basic software, various optional packages can enhance the capabilities of the system. Some of the capabilities described in this document require optional software. See your Account Representative for more information. The basic software is a prerequisite for all the optional packages.

2 —Telephone Features

Telephones

DEFINITY ECS telephones fall into three basic families—Analog, Digital Communications Protocol (DCP), and BRI. These terms describe how each type of telephone communicates with the DEFINITY ECS switch. These families of telephones are designed to accommodate the types of communications various users require. All telephones have touch-tone dialing and the message-waiting lamp for notification of messages.

Telephones for the Global Marketplace

With help from our many global customers, Lucent Technologies has developed the 8400, 9400, 6400 and 6200 series telephones to meet the demand for two-wire telephones in the global marketplace.

8400-Series Telephones

The 8400 digital telephones are versatile two-wire/four-wire Digital Communications Protocol (DCP) telephones with new styling that offer new flexibility and cost savings. They automatically detect whether they are plugged into a two-wire or four-wire digital line circuit card. This is a significant benefit because it provides an easier transition to either a two-line or a four-line environment, therefore reducing wiring expenses and installation adjustments. It also allows you to save space inside the cabinet by using 24-port two-wire boards in place of 16-port four-wire boards.

9400-Series Telephones

The 9400 digital telephones, also known as Europhones, provide inexpensive support for two-wire installations, while still providing a European design. Three models of the 9400 telephones are available in gray and cream white. The 9403, 9410B, 9410D, and 9434 telephones are similar in design and features to the 8400 series.

6400-Series Telephones

The 2-wire, DCP 6400 digital telephones are similar to the 8400 and 9400 telephones, and feature new styling and a pullout instruction card. The 6400 telephones also include the following additional features:

- Date and time display.
- A feature button which allows switchhook control of a headset.
- *Group Listen* capability, which allows you to use your handset or headset normally while others in the room listen in via speakerphone. This 2-way handset, 1-way speaker mode allows you to serve as a spokesperson for a group.
- *Telephone Self Administration* capability, which allows you to program feature buttons on the telephone yourself.

6400 Tip/Ring Interface Module

This module provides a two wire analog interface for the 6400 DCP telephones. This will allow the operation of an analog adjunct to be independent of the digital telephone's extension for the use of FAX machines or modems without compromising the user's voice extensions.

6200-Series Analog Telephones

The 6210 and 6220 analog telephones are designed to take advantage of the many features offered by the DEFINITY ECS. They offer the following features.

- Message light
- Flash and redial buttons
- Hold button and hold light
- Handset volume control
- Data jack (for connecting a modem or similar device)
- Personalized ringing, speakerphone button and light, and programmable dialing buttons (6220 only)

IP SoftPhones

IP SoftPhones extend the level of DEFINITY services. They turn a PC or a laptop into an advanced telephone. Users can place calls, take calls, and handle multiple calls on their PCs. For a discussion of the types of SoftPhones available with IP Solutions for DEFINITY ECS, see "IP SoftPhones" on page 58.

Wireless Handsets for X-Station Mobility

X-Station Mobility allows remote users to access switch features. It allows wireless telephones remoted over a trunk interface to be controlled by the DEFINITY ECS as if the telephones were directly connected to the switch.

Teleconferencing Products

Quorum A-28 Conference Bridge

The Quorum A-28 Conference Bridge is a microprocessor-controlled analog bridge. It allows you to activate a multipoint connection of switched and private-line circuits to create a single conversation. You can set up, monitor, and control your own teleconferences through your DEFINITY ECS.

The Quorum A-28 Conference Bridge allows you to connect up to 28 different people on a single conference call. Or you can set up multiple conferences — up to four conferences with seven participants each.

The bridge also provides a lecture mode that you can use for one-way broadcasts. Using the Mute feature, participants can be placed in a listen-only mode, allowing one speaker to address the rest of the conference.

SoundStation Audioconferencing Systems

Lucent Technologies's SoundStation and SoundStation EX Audioconferencing Systems enable a group of people in a conference room to share their conversation with others through a telephone connection. The Soundstation equipment permits natural conversation among many people — whether strong or soft, or from a standing or sitting position.

SoundStation

The SoundStation has three microphones and a digitally tuned speaker that provide 360-degree coverage, whether you use the system in an office or a conference room. It connects to an analog telephone line. The built-in keypad includes a mute button and a flash key. An additional port allows you to connect the speakerphone to a tape recorder.

SoundStation EX

The SoundStation EX includes all the features and functions of the SoundStation. It accommodates larger conferences by including two palm-size external microphones that can be positioned up to six feet (1.8 m) on either side of the center console. An optional wireless microphone is available for stand-up presenters. See Figure 7 for an illustration of the SoundStation EX with External Microphones.

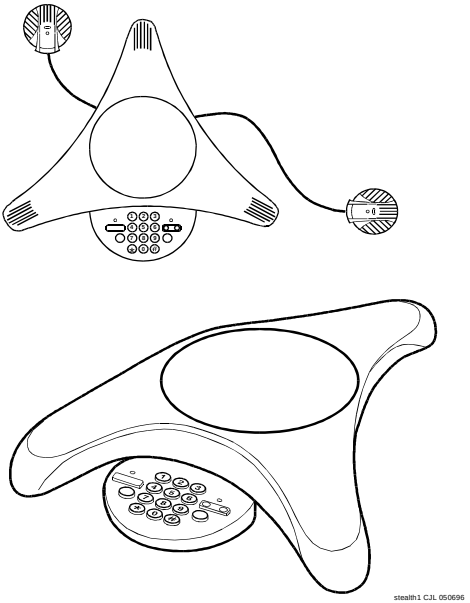


Figure 7. Soundstation EX with External Microphones

Telephone Features

Handling Incoming Calls

Automatic Hold

Allow Attendants and multi-function telephone users to alternate easily between two or more calls. For example, with automatic hold, selection of a second call automatically puts the active call (if any) on hold and makes the second call active. This feature can be activated on a system-wide basis only. When automatic hold is not activated, the selection of the second call drops the first call.

Hold

Allows you to disconnect from a call temporarily, use your telephone for other call purposes, and then return to the original call.

Long Hold Recall

Visual and audible warnings are sent to the telephone where a call has been on hold past a specified period of time. Both visual and audible warnings are used if the telephone is on-hook. If the telephone is off-hook, a "priority ring" is used. The call is identified as a long-held call by the letters "hc" on the display. This is an optional feature at the system level.

Transfer

Allows telephone users to transfer trunk or internal calls to other telephones within the system without attendant assistance. This feature provides a convenient way to connect a party with someone better qualified to handle the call. Single-line telephone users momentarily flash the switchhook or press the Recall button, dial the desired extension, and hang up. Multi-appearance telephone users press the Transfer button, dial the desired extension number, and press the Transfer button again. This is an optional feature at the system level.

Pull Transfer

Allows *either* the party who was originally called *or* the party to whom the held call will be transferred to complete the transfer. This is a convenient way to connect a party with someone better qualified to handle the call. Attendant assistance is not required and the call does not have to be redialed. It interfaces with satellite workstations via TGU/TGE trunks and is always available for calls that use TGU/TGE trunks.

Trunk-to-Trunk Transfer

Allows the attendant or telephone user to connect an incoming trunk call to an outgoing trunk call. This feature is particularly useful when a caller outside the system calls a user or attendant and requests a transfer to another outside number. For example, a worker, away on business, can call in and have the call transferred elsewhere. The system assures that incoming Central Office trunks without Disconnect Supervision are not transferred to outgoing trunks or other incoming Central Office trunks without Disconnect Supervision.

Transfer — Outgoing Trunk to Outgoing Trunk

Allows a user or attendant to initiate two or more outgoing trunk calls and then transfer the trunks together. The transfer operation removes the original user from the connection and conferences the outgoing trunks. Alternatively, the controlling party can establish a conference call with the outgoing trunks and then drop out of the conference, leaving only the outgoing trunks on the conference. This is an optional enhancement to Trunk-to-Trunk Transfer and requires careful administration and use. DCS Trunk Turnaround may be a safer alternative to this feature.

Transfer Upon Hang-Up

Provides you with the ability to transfer a call by hanging up instead of having to press the Transfer button a second time. You would press the Transfer button, dial the number the call is being transferred to and then hang up. This is an optional feature at the system level. You will still be able to transfer a call by pressing the Transfer button a second time.

Transfer Recall

Returns the unanswered transfer calls back to the person who transferred the call. Transfer Recall uses a priority alerting signal, and the display on the telephone shows "rt", which indicates a returned call from a failed transfer operation.

Abort Transfer

Allows a user to abort a transfer attempt by pressing a non-idle line appearance. The call being transferred would be taken off a transfer-type hold and be put on a traditional hold. The transfer will also be aborted when you hang up (going on-hook), unless Transfer Upon Hang-Up is activated on the switch. This is an optional feature at the system level.

Conference

The Conference button allows multiappearance telephone users to make up to six-party conference calls without attendant assistance. This feature also allows single-line telephone users to make up to three-party conference calls without attendant assistance.

Abort Conference on Hangup

When you punch the conference button and for any reason you hang up before you complete the conference, you will cancel the conference. The original call that was put on soft-hold will now be on hard-hold.

Call Park

Allows you to put a call on hold and then retrieve a call from any other telephone on the system. This is helpful when you are on a call and need to go to another location for information. It also allows you to answer a call from any telephone after being paged by a telephone user or an attendant.

Misoperation Handling

NOTE:
This feature is required only in France and Italy, but it can be used at any location where the feature has been turned on.

Defines how calls are handled when a misoperation occurs. A misoperation is when calls are left on hold when the controlling station goes on hook.

For example, a misoperation can occur under either of the following conditions:

- If you hang up prior to completing a feature operation (in some cases, hanging up completes the operation, as in call transfer). If, for example, you place a call on hold, begin to transfer the call, dial an invalid extension number, and then hang up, that's a misoperation.
- When the system enters night service while attendant consoles have calls on hold.

The system administrator can alter the standard Misoperation Handling to ensure that an external caller is not left on hold indefinitely, or dropped by the system after a misoperation with no way to reach someone for help.

Manual Message Waiting

Allows multi-appearance telephone users to light the status lamp associated with the manual Message Waiting button at another multi-appearance telephone. They do this by simply pressing a button on their own telephone. This feature can be administered only to

pairs of telephones such as a secretary and an executive. The secretary might press the button to signal to the executive that a call needs answering or someone has arrived for an appointment. The executive might use the button to indicate that he or she should not be disturbed.

Internal Automatic Answer

Allows specific telephones to answer incoming internal calls automatically. This feature is intended for use with telephones that have speakerphones or headsets. You simply press an Internal Automatic Answer feature button, and calls are automatically answered when the telephone is idle. Internal and Distributed Communications System (DCS) calls can be answered using Automatic Answer, but only attendants can use Automatic Answer to answer external calls directed to the attendant.

Recall Signaling

Recall Signaling allows the user of an analog station to place a call on hold, use the voice terminal for other call purposes, and then return to the original call.

Local Call Timer Automatic Start/Stop

Automatically starts the local timer of a 6400 series telephone when a call is received. The timer is stopped automatically when a call is ended. When a call is placed on hold the timer continues to run, but is not displayed. When the call comes off hold, the total elapsed call-time displays.

Call Redirection

Call Forwarding

Call Forwarding provides four functions:

- Call Forwarding All Calls — Allows calls to be forwarded to an internal extension, external (off-net) number, an attendant, or an attendant group.
- Call Forwarding Override — Allows the user at the forwarded-to extension to override Call Forwarding and either initiate a call or transfer a call back to the forwarded-from extension.
- Call Forward Busy/Don't Answer — Allows calls to be forwarded when the called extension is busy or when the call is not answered after an administrable interval. If the extension is busy, the call forwards immediately. If the extension is not busy, the incoming call rings the called extension, then forwards only if it remains unanswered longer than the administered interval.

- Call Forwarding Off Net — Allows calls forwarded off net to be tracked for busy or no-answer conditions. The system brings the call back for further call-coverage processing if specified conditions are met. This feature is particularly useful for Telecommuters, who can have their on-site office calls forwarded to their home offices.

Call Coverage

Call Coverage provides automatic redirection of calls that meet specified criteria to alternate answering positions in a Call Coverage path. A coverage path can include any of the following: a telephone, an attendant group, a Uniform Call Distribution (UCD) hunt group, a Direct Department Calling (DDC) hunt group, an Automatic Call Distribution (ACD) hunt group, a voice messaging system, or a Coverage Answer Group (CAG) established to answer redirected calls.

In addition to redirecting a call to a local answering position, you can administer Call Coverage to:

- Redirect calls based on time-of-day
- Redirect calls to a remote location
- Allow users to change back and forth between two lead-coverage paths from either an on- or off-site location

Coverage Callback

Allows a covering user to leave a message for the called party to call back the person who called.

Coverage Incoming Call Identification

Allows multi-appearance telephones users without a display in a Coverage Answer Group to identify an incoming call to that group.

Go to Cover

Allows users who call another internal extension to send the call directly to coverage.

Send All Calls

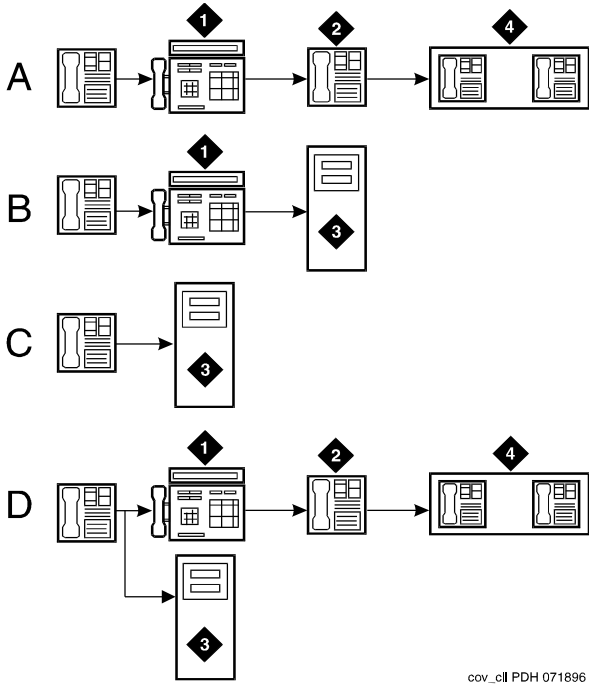
Allows users to temporarily direct all incoming calls to coverage regardless of the assigned call-coverage redirection criteria. Covering users can temporarily remove their voice terminals from the coverage path. The feature is activated and deactivated via a button or access code.

Consult

Allows a covering user, after answering a call received through Call Coverage, to call the called party for private consultation. Consult can be used to let a covering user ask the principal if they want to speak with the calling party.

Reset Shift Call

If a call number is busy and doesn't have coverage or the called number and the coverage are both busy, you have an opportunity to replace the last digit that was entered. This allows you to call another extension without having to hang up and redial. Reset Shift Call is a feature that is active for station to station (internal) calls and for Private Network calls. The Private Network trunks must signal busy using out-of-band signaling.



A	External Call: Active, Busy, Don't Answer	1	Secretary
B	Internal Calls: Cover All	2	Clerk
C	Internal Call: Active, Busy, Don't Answer	3	AUDIX Voice Messaging
D	Internal Calls: Send All Calls	4	Message Center Group

Figure 8. Typical DEFINITY ECS Call Coverage Options

Night Service

There are five Night Service features:

- Hunt Group Night Service allows an attendant or a split supervisor to assign a hunt group or split to Night Service mode. All calls for the hunt group then are redirected to the hunt group's designated Night Service extension. When a user activates Hunt Group Night Service, the associated button lamp lights.
- Night Console Service directs all calls for primary and daytime attendant consoles to a night console. When a user activates Night Console Service, the Night Service button for each attendant lights and all attendant-seeking calls (and calls waiting) in the queue are directed to the night console. To activate and deactivate this feature, the attendant typically presses the Night button on the principal attendant console or designated console.
- Night Station Service directs incoming calls for the attendant to designated extensions. Attendants can activate Night Station Service by pressing the Night button on the principle console if there is not an active night console. If the night station is busy, calls (including emergency attendant calls) receive a busy tone. They do not queue for the attendant.
- Trunk Answer from Any Station allows telephone users to answer all incoming calls to the attendant when the attendant is not on duty and when other voice terminals have not been designated to answer the calls. The incoming call activates a gong, bell, or chime and a voice-terminal user dials an access code to answer the call.
- Trunk Group Night Service allows an attendant or a designated telephone user to individually assign a trunk group or all trunk groups to the night service mode. Specific trunk groups individually assigned to the service are in Individual Trunk Night Service Mode. Calls coming into these trunk groups are redirected to designated night service extensions. Incoming calls on other trunk groups are processed normally.

Enhanced Night Service

DEFINITY ECS informs a Voice Mail System (VMS) that it is in Night Service, allowing the VMS to perform different actions and call handling for out-of-hours operation. For example, the VMS may be administered to provide recorded announcements after hours. The enhancement is made to the Mode Code Voice Mail Interface.

Bridged Call Appearance — Multi-Appearance Telephone

Allows calls made to or from a primary telephone user's extension number to be handled from more than one telephone. A bridged call appearance is set up by administering a primary extension and the button number associated with it on a multi-lamp button on another telephone. One way this feature is most often used is by secretaries or assistants who answer or handle calls to the primary extension (an executive, for example). When the primary extension receives a call, the bridged call appearance flashes or rings on all telephones administered with this feature. The call can be answered by anyone having a telephone with this feature and handled as if the primary extension user was answering it. The maximum number of bridged appearances has been expanded from 25 to 64 with DEFINITY ECS R8.

Bridged Call Appearance — Single-Line Telephone

Allows single-line telephones users to have a bridged appearance on a multi-appearance telephone.

Temporary Bridged Appearance

Allows multi-appearance telephone users in a terminating extension group or personal central office line group to bridge onto an existing group call. If a call has been answered using the Call Pickup feature, the originally called party can bridge onto the call. This feature also allows a called party to bridge onto a call that redirects to coverage before the called party can answer it.

Privacy — Manual Exclusion

Allows multi-appearance telephone users to keep other users with appearances of the same extension number from bridging onto an existing call. Exclusion is activated by pressing the Exclusion button on a per-call basis.

Privacy-Auto Exclusion

When the Class of Service is set for the Automatic Exclusion option, the feature is activated when you take your telephone off-hook. The feature can be deactivated when you push the Exclusion button before dialing a call or during a call. An excluded call that is on hold can be taken off hold by any telephone that has a bridged appearance of the telephone that put the call on hold.

Call Pickup

Along with Directed Call Pickup, allows you to answer calls for other telephones within your specified call pickup group. Directed Call Pickup allows you to pick up any call on the DEFINITY ECS system. With this feature, you do not have to leave your telephone to answer a call for a nearby telephone. You simply dial an access code or press a Call Pickup button.

Group Call Pickup

Allows you to dial a Feature Access Code (FAC) and a Pickup Group Number to answer a call from a different group. For example, Marketing would be able to pickup calls in the Sales group when the Sales group is unavailable. This feature is ideal for offices that are not divided by partitions and generally have the departments on the same floor.

Terminating Extension Group

Allows an incoming call to ring (either audible or silent alerting) as many as four telephones at the same time. Any user in the group can answer the call. Any telephone can be administered as a group member. Only a multi-appearance telephone can be assigned a feature button with an associated status lamp, however. The feature button allows the user to select a Terminating Extension Group call appearance for answering or bridging onto an existing call but not for call origination. For example, a department in a large store might have three telephones. Anyone in the department can answer the call. The salesperson most qualified to answer the call can bridge onto the call.

Station Hunting

Routes calls made to a busy extension to another extension. To use Station Hunting, you create a station hunting chain that governs the order in which a call routes from one extension to the next when the called extension is busy. Each extension in the chain links *to* only one subsequent extension. An extension may be linked *from* any number of extensions, however.

Station Hunt Before Coverage

This feature changes the interaction that occurs between station hunting and call coverage. Station Hunt before Coverage causes a call going to a busy station to go through a station hunting process before going to coverage. If all the stations in the Hunt group are busy, the call will go to the coverage path.

Circular Station Hunting

This new hunt group type is an alternative to the "ddc" or "hot-seat" algorithm in a hunt group. DEFINITY ECS keeps track of the last extension in the hunt group that received a call. When another incoming call arrives, it is sent to the next idle extension, bypassing the extension that had received the previous call. The first extension in the hunt group will no longer be the busiest telephone while the others in the group are sitting idle.

Speed/Convenience Calling

Abbreviated Dialing

Provides lists of stored numbers you can use to:

- Place local, long-distance, and international calls
- Activate features
- Access remote computer equipment

You simply dial the list number and the one-, two-, or three-digit number associated with the telephone number you want. The number is then automatically dialed by the system. A frequently called number can be stored on an abbreviated dialing button that you need only press once to make the call.

Abbreviated Dialing Labeling

Labeling of Abbreviated Dialing (AD) Buttons on Softkeys allows users of 8400 and 6400 series display telephone sets to administer labels for the AD buttons that appear on their softkeys. These personalized labels appear on the menu display.

Abbreviated Dialing On-Hook Programming

On-Hook Programming allows users of 8400 and 6400 series telephone sets with enabled speakers to access the programming mode without going off-hook during available call appearances. Signaling changes from DTMF to the S-channel, allowing the use of a longer (60 seconds) time-out period. Signaling will remain DTMF and the current time-out period of 10 seconds will still apply to non-display telephone sets.

Enhanced Abbreviated Dialing

Supplements Abbreviated Dialing by providing one enhanced number per system. Enhanced number lists can contain any number or dial access code. System Administrators designate privileges for group number lists, system number lists and

enhanced number lists. With privileged lists, users can access otherwise-restricted numbers (e.g., Stations without long-distance access can be programmed to access specified long-distance numbers).

Active Dialing

6400 series telephone sets have a dialing option where the set will send S-channel button codes when the user presses a number on the dial pad when on-hook.

Telephone Self-Administration

Allows you to program feature buttons on 6400-series telephones yourself.

Automatic Callback

Allows internal users who placed a call to a busy or unanswered internal telephone to be called back automatically when the called voice terminal becomes available.

When a user activates Automatic Callback, the system monitors the called telephone. When the called telephone becomes available to receive a call, the system originates the Automatic Callback call. The originating party receives priority ringing. The calling party then lifts the handset and the called party receives the same ringing provided on the original call.

Ringback Queuing

Places calls in an ordered queue (first in, first out) when all trunks are busy. The telephone user who is trying to make a call is automatically called back when a trunk becomes available, and hears a distinctive three-burst signal when called back.

Last Number Dialed

Allows you to automatically redial the last number dialed. The system saves the first 24 digits of the last number dialed, whether the call attempt was manually dialed or dialed using Abbreviated Dialing. When you press the Last Number Dialed button or dial the Last Number dialed feature access code, the system places the call again.

Remote Access

Permits authorized callers from remote locations to access the system via the public network and then use its features and services.There are a variety of ways of accessing the feature. After gaining access, you hear a system dial tone, and, for system security, may be required to dial a barrier code.

Recorded Telephone Dictation Access

Allows telephone users, including Remote Access and incoming tie trunk users, to access dictation equipment. The dictation equipment is accessed by dialing an access code or extension number. The start/stop function can be voice or dial controlled. Other functions such as initial activation and playback are controlled by additional dial codes.

Emergency Access to the Attendant

Provides for emergency calls to be placed to an Attendant. These calls can be placed automatically by the system or can be dialed by system users. Emergency access calls can receive priority handling by the Attendant.

Crisis Alert

Crisis Alert uses both audible and visual alerting to notify designated extensions when an emergency call is made. Audible alerting sounds like an ambulance siren. Visual alerting flashes the CRSS-ALRT button lamp and the display of the caller's name and extension. Crisis Alert's display of the origin of the emergency call enables the attendant or other user to direct emergency-service response to the caller.

If an emergency call is made while another crisis alert is still active, the incoming call will be placed in the queue. If the system is administered so that all users must respond, then every user must respond to every call, in which case the calls are not necessarily queued in the order in which they were made. If the system is administered so that only one user must respond, the first crisis alert remains active at the phone where it was acknowledged. Subsequent calls are queued to the next available station in the order in which they were made.

Crisis Alert can also send notification of an emergency call to a digital pager. In this case, it sends a message of 7 to 22 digits to the pager and displays a crisis alert code, an extension and room number, and a main number (if one is entered). The person paged thus knows the origin of the emergency call and can direct emergency-service response to the appropriate location. To use Crisis Alert with a digital pager, the system is administered so that at least one digital set has a CRSS-ALRT button and the Alert Pager field is set to **y**. Any station with a CRSS-ALRT button and a pager receives the correct alert.

Manual Originating Line Service

Connects single-line telephone users to the attendant automatically when the user lifts the handset. The attendant number is stored in an Abbreviated Dialing list. When the telephone user lifts the handset, the system automatically routes the call to the attendant using the Hot Line Service feature.

Trunk Flash

Trunk Flash allows a feature or function button on a multifunction telephone or attendant console to be assigned as a Flash button. Pressing this button while connected to a trunk (which must have been administered to allow Trunk Flash) causes the system to send a flash signal out over the connected trunk.

Trunk Flash enables multifunction voice terminals to access central office customized services that are provided by the Central Office to which DEFINITY ECS is connected. These services are electronic features, such as conference and transfer, that are accessed by a sequence of flash signal and dial signals from the DEFINITY System station on an active trunk call. The Trunk Flash feature can help to reduce the number of trunk lines connected to the DEFINITY system. “Digit 1 as Flash” as used in Italy and the United Kingdom will not serve as the flash button in this application.

Special Dial Tone

Provides the ability to play a Special Dial Tone whenever an analog set is not able to receive calls. When such conditions as Call Forward All Calls, Call Forward Busy/NA, Send All Calls or Do Not Disturb are activated on a telephone set, a Special Dial Tone lets you know that you cannot receive any calls.

Telephone Displays

Voice Terminal Display

Provides multi-appearance telephone users with updated call and message information. This information is displayed on a display-equipped telephone. The information displayed depends upon the display mode selected by the user. Information that allows personalized call answering is available on many calls.

Users may select any of the following as the display message language: English (default), French, Italian, or Spanish. In addition, messages can be administered on the system in a fifth language. The language for display messages is selected by each user.

ICLID on Analog Trunk

In the US and Japan, the user's terminal displays calling party information. Name and calling number are available from the US central offices; only the calling number is available from central offices in Japan. This feature may be used in countries that comply with either US or Japanese requirements. The display of name and number will work with all DEFINITY digital voice terminals (DCP and BRI) equipped with a 40-character or a 32-character alphanumeric display.

Enhanced Voice Terminal Display

The Enhanced Voice Terminal Display feature allows you to choose the character set that you want to see in DEFINITY ECS softkeys and display terminals. In addition to the standard Roman character set, you can choose either the Katakana or characters used for most European languages.

Administrable Language Displays

Allows the messages that appear on telephone display units to be shown in the language spoken by the user. These messages are available in English (the default), French, Italian, Spanish, or one other user-defined language. The language for display messages is selected by each user. The feature requires 40-character display telephones.

Directory

Allows users with display-equipped telephones to access the system database, use the touch-tone buttons to enter a name, and retrieve an extension number from the system directory. The directory contains the names and extensions assigned to all telephones on the system.

Group Communication

Conference — Terminal

Allows multi-appearance telephone users to set up six-party conference calls without attendant assistance. Single-line telephone users can set up three-party conference calls without attendant assistance.

Intercom — Automatic

Allows two users to talk together easily. Calling users press the Automatic Intercom button and lift the handset. The called user receives a unique intercom ring and the intercom lamp, if provided, flashes. With this feature, users who frequently call each other can do so by pressing one button instead of dialing an extension number.

Intercom — Dial

Allows multi-appearance telephone users to easily call others within an administered group. The calling user lifts the handset, presses the Dial Intercom button, and dials the one- or two-digit code assigned to the desired party. The called user's telephone rings, and intercom lamp, if provided, flashes. With this feature, a group of users who frequently call each other can do so by pressing one button and dialing a one- or two- digit code instead of dialing an extension number.

Manual Signaling

Allows one user to signal another user. The receiving user hears a two-second ring. The signal is sent each time the button is pressed by the signaling user. The meaning of the signal is prearranged between the sender and the receiver. Manual Signaling is denied if the receiving telephone is already ringing from an incoming call.

Group Listen

Simultaneously activates your speakerphone in listen only mode and your handset or headset in listen and speak mode. This allows you to serve as spokesperson for a group. You can participate in a conversation while everyone else in the room is listening to what is said.

Whisper Page

Allows an assistant or colleague to bridge onto your telephone conversation and give you a message without being heard by the other party or parties you are talking to. Whisper Page works only on certain types of telephones.

Loudspeaker Paging Access

Provides attendants and telephone users dial access to voice paging equipment. As many as nine paging zones can be provided by the system and one zone can be provided that activates all zones at the same time. (A zone is the location of the loudspeakers — for example, conference rooms, warehouses, or storerooms.) A user can activate this feature by dialing the trunk access code of the desired paging zone, or the access codes can be entered into Abbreviated Dialing Lists. Once you have activated this feature, you can simply speak into the handset to make the announcement.

Deluxe Loudspeaker Paging Access (called Deluxe Paging) provides attendants and telephone users with integrated access to voice-paging equipment and Call Park capabilities. When you activate Deluxe Paging, the call is automatically parked. The parked call returns to the parking user with distinctive alerting when the time-out interval expires.

Code Calling Access

Allows attendants, users, and tie trunk users to page with coded chime signals. This feature is helpful for users who are often away from their telephones or at a location where a ringing telephone might be disturbing.

Special Ringing

Distinctive Ringing

Rings or activates alerting on your telephone in such a way that you are aware of the type of incoming call before answering it. This feature operates in a Distributed Communication System (DCS) environment the same as it does within a single system.

By default, internal calls are identified by a 1-burst ringing pattern, external calls by a 2-burst ringing pattern, and priority calls by a 3-burst ringing pattern. You can administer these patterns, however.

Personalized Ringing

Allows users of certain telephones to uniquely identify their own calls. Each user can choose one of a number of possible ringing patterns. The eight ringing patterns are tone sequences consisting of different combinations of three tones. With this feature, users working closely in the same area can each specify a different ringing pattern in order to better identify their own calls.

Priority Calling

Allows you to ring another telephone with a distinctive signal that tells the called party the incoming call requires immediate attention. The called party can then handle the call accordingly. You activate priority calling by Dialing a Priority Calling access code or pressing a feature button, followed by the extension number. You can use Priority Calling only if your telephone has been administered with the required class of service.

Voice Terminal Alerting Options

Provides multi-appearance telephone users with different ringing patterns. This feature primarily affects audible ringing for calls directed to telephones that are off hook, or calls directed to idle and active CALLMASTER telephones.

Ringing — Abbreviated and Delayed

Allows you to manually or automatically assign one of four ring types to each call appearance on a telephone. Whatever treatment you assign to a call appearance is automatically assigned to each of its bridged call appearances.

Ringer Cutoff

Allows the user of a multi-appearance telephone to turn audible ringing signals on and off. Visual alerting is not affected by this feature. When this feature is enabled, only Priority (three-burst) ring, Redirect Notification, Intercom ring, and Manual Signaling ring at the telephone. Internal and external calls do not ring.

Multiappearance Preselection and Preference

Provides options for placing or answering calls on selected call appearances. Ringing Appearance *Preference* automatically connects you to the incoming ringing call when the user picks up the handset. *Idle Appearance Preference* automatically connects you to an idle appearance. *Preselection* allows the user to manually select an appearance. Preselection is used, for example, when you want to reconnect with a held call or activate a feature. Preselection can be used with a feature button. For example, if you press an Abbreviated Dialing button, the call appearance is automatically selected and, if you pick up the handset within five seconds, the call is automatically placed. The Preselection option overrides both of the other preference options.

Messaging

Leave Word Calling

Allows internal system users to leave a short preprogammed message (usually “Call” with the calling user’s name, extension number, and the time of the call) for other internal users. When the message is stored on the DEFINITY ECS, the Message Lamp on the called telephone automatically lights. Leave Word Calling messages can be retrieved using a telephone display, Voice Message Retrieval, or AUDIX. Messages may be retrieved in English, French, Italian, Spanish, or a user-defined language.

Audible Message Waiting

Places a stutter at the beginning of the dial tone when a telephone user picks up the telephone. The stutter dial tone indicates that the user has a message waiting. This feature is particularly useful for visually impaired people who may not be able to see a

message light. It is often used with telephones that have no Message Waiting Lights. Audible Message Waiting may not be available in countries that restrict the characteristics of dial tones provided to users.

Voice Messaging and Call Coverage

Often an AUDIX system is set up as the last point on a call-coverage path, as in Figure 8. A secretary or colleague who answers a redirected call intended for you can also transfer the caller to your AUDIX mailbox. The caller may prefer to leave voice-mail for you if the message is personal, lengthy, or technical.

Many other options are available. For example, a caller can redirect a call from the AUDIX system to an attendant. Or the caller can transfer to another extension instead of leaving a message. You can even have the AUDIX automated attendant answer all calls to the company and send calls to various extensions. In this case, callers are instructed to enter keypad commands to direct the call.

Voice Message Retrieval

Allows telephone users, remote access users, and attendants to retrieve Leave Word Calling and Call Coverage voice messages. It can be used to retrieve a user's own messages or messages for another user. However, a different user's messages can be retrieved only by a user at a telephone or attendant console in the coverage path, by an administered system-wide message retriever, or by a remote-access user when the extension and associated security code are known. The system restricts unauthorized users from retrieving messages.

Message-Retrieval Options

With the Message-Waiting Lamp on their telephones, employees always know when they have messages. Messages can be retrieved in a variety of ways, such as:

- Display retrieval — Users having digital telephones with displays or a personal computer integrated with a telephone can display messages.
- Speak-to-Me — Using any touch-tone telephone, employees can dial Speak-to-Me and hear a synthesized voice read their messages over the telephone.

These message-retrieval options can be assigned to users individually.

Demand Print

Allows you to print your undelivered messages without calling the Message Center.

Voice Messaging Systems

The Lucent Technologies voice messaging solutions include:

- DEFINITY AUDIX
- INTUITY AUDIX

Voice Response solutions include:

- CONVERSANT Voice Information System V7

Telecommuting

Remote Call Coverage/Call Forwarding Off-Net

Remote Call Coverage and Call Forwarding Off-Net allow calls to be redirected to a remote location. This allows you to have calls placed to your on-site office redirected to your home office. You can administer the system to either monitor calls and bring them back for additional processing if not answered or to leave calls at the remote (off-net) location.

Extended User Administration of Redirected Calls (Telecommuting Access)

Extended User Administration of Redirected Calls (also called Telecommuting Access) allows you to change the lead call coverage path or forwarding extension from any on-site or off-site location. Thus you can change the path or extension from your home office, for example.

Off-Premises Station

A trunk-data module connects off-premises private-line trunk facilities and DEFINITY ECS. The trunk-data module converts between the RS-232C and the DCP, and can connect to DDD modems as the DCP member of a modem pool.

See also, “Call Redirection” on page 22.

See also, “Call Vectoring” on page 97.

Personal Station Access

Allows you to transfer your telephone station preferences and permissions to any other compatible telephone. This includes the definition of terminal buttons, abbreviated dial lists, and Class of Service and Class of Restrictions permissions. It can be used on-site

or off-site (with DEFINITY Extender). Personal Station Access can also be used to prevent unauthorized calls from your phone when you are away from your desk. You would disassociate your desk telephone from your extension number when you are away from your desk and re-associate your desk telephone number with your extension number when you return. Personal Station Access has several telecommuting applications. For example, several telecommuting employees can share the same office on different days of the week. The employees can easily and remotely make the shared telephone “theirs” for the day. Remote use requires DEFINITY Extender.

Station Self Display

Station Self Display shows the extension number of the telephone set when a user either dials the Feature Access Code while off-hook or depresses the "Inspect" button when on-hook. The dialed number will be displayed once the user starts to dial. This feature is helpful to people who move from one desk to another while they are working. This feature is also used by maintenance personnel to ensure that an extension number is correctly administered.

3 —Attendant (Operator) Features

Attendant (Operator)

A person at a console who provides personalized service for incoming callers and voice-services users by performing switching and signaling operations.

Attendant Console

A digital call-handling station with push-button control used not only to answer and place calls, but also to manage and monitor some system operations.

The Attendant Display shows call-related information that helps the attendant to operate the console. Also shows personal service and message information. Information is shown on the alphanumeric display on the attendant console. Attendants may select one of several available display message languages: English, French, Italian, or Spanish. In addition, your company may define one additional language for use by users and attendants on their display.

DEFINITY PC Console

Lucent Technologies DEFINITY PC Console allows your Attendants to handle incoming calls efficiently by personal computer. Using the familiar Microsoft Windows graphical interface, the Attendants can easily keep track of how long callers have been on hold and who they are waiting for. Attendants can monitor up to six calls at once. They need not fumble with pen and paper when handling calls, as they can make notes on their computers about what each caller needs. All this contributes to make a favorable first impression with your customers. Having the call processing software on the same computer with spreadsheet, word processing, or other software allows the attendants to stay productive between calls.

The PC Console is easily customized, so even if attendants from different shifts share the same computer, they can each preserve their preferences in the call processing environment. The PC Console is available in English, Parisian French, Latin American Spanish, German, Dutch, Italian and Portuguese. It will be available in Italian in the fall of

1997. If a Spanish-speaking Attendant takes over for a French-speaking attendant, for example, a single press of a button converts all labels, error messages and online help to Spanish.

Attendant Features

Call Handling

Listed Directory Number

Allows outside callers to access your attendant group in two ways, depending on the type of trunk used for the incoming call. You can allow attendant group access via incoming direct inward dial trunks, or you can allow attendant group access via incoming Central Office) and foreign exchange trunks.

Call Waiting

Allows an Attendant to let a single-line telephone user who is on the phone know that a call is waiting. The Attendant is then free to answer other calls. The Attendant hears a call waiting ringback tone and the busy telephone user hears a call waiting tone. This tone is heard only by the called telephone user.

Calling of Inward Restricted Stations

A telephone with a Class of Restriction that is inward restricted cannot receive public network, attendant-originated, or attendant-extended calls. This feature allows you to override this restriction.

Priority Queue

Places incoming calls to the Attendant in an orderly queue when these calls cannot go immediately to the Attendant. This feature allows you to define twelve different categories of incoming attendant calls, including emergency calls, which are given the highest priority.

Override of Diversion Features

Allows an Attendant to bypass diversion features such as Send All Calls and Call Coverage by putting a call through to an extension even when these diversion features are on. This feature, together with Attendant Intrusion, can be used to get an emergency or urgent call through to a telephone user.

Backup Alerting

Notifies backup Attendants that the primary Attendant cannot pick up a call. It provides both audible and visual alerting to backup stations when the attendant queue reaches its queue warning level. When the queue drops below the queue warning level, alerting stops. Audible alerting also occurs when the attendant console is in night mode, regardless of the Attendant queue size.

Timed Reminder and Attendant Timers

Automatically alerts the Attendant after an administered time interval for the following types of calls: extended calls to be answered or waiting to be connected to a busy single-line telephone, one-party calls placed on hold on the console, and transferred calls that have not been answered after transfer. Timed Reminder informs the Attendant that a call requires additional attention. After the Attendant reconnects to the call, the user can either choose to try another extension number, hang up, or continue to wait. DEFINITY ECS supports a variety of administrable attendant timers for use in a variety of situations.

Privacy — Attendant Lockout

Prevents an Attendant from re-entering a multiple-party connection held on the console unless recalled by a telephone user. This feature is administered on a system-wide basis. It is either activated or not activated.

Intrusion (Call Offer)

Allows an Attendant to enter an existing call to inform the person being called about a message or another call.

Release Loop Operation

Allows the Attendant to hold a call at the console if the call cannot immediately go through to the person being called. A timed reminder begins once the call is on hold. If the call is not answered within the allotted time, the call returns to the queue for the Attendant. Timed reminders attempt to return the call to the Attendant who previously handled it. Only when the original Attendant is unavailable are calls returned to the queue.

Attendant Vectoring

Provides you with the flexibility to manage Attendant group voice calls. When this feature is turned on, you will be able to program how you want your Attendant group calls processed. Attendant vectoring can be used by itself or in combination with QSIG Centralized Attendant Service.

Serial Calling

Enables an Attendant to transfer trunk calls that return to the same Attendant after the called party hangs up. The returned call can then transfer to another station within the switch. This feature is useful if trunks are scarce and Direct Inward Dialing services are unavailable. An outside caller may have to redial often to get through because trunks are so busy. Once callers get through to an Attendant they can use the same line into the switch for multiple calls. The Attendant's display shows if an incoming call is a serial call.

Conference

Allows an Attendant to set up a conference call for as many as six conferees, including the Attendant. Conferences from inside and outside the system can be added to the conference call.

Making Calls

Auto Start and Don't Split

Auto Start allows the Attendant to make a telephone call without pushing the start button first. If the Attendant is on an active call and presses digits on the keypad, the system automatically splits the call and begins dialing the second call. The Don't Split feature deactivates the Auto Start feature and allows the sending of touch tones over the line for the purposes of such things as picking up messages.

Auto-Manual Splitting

Allows an Attendant to announce a call or consult privately with the called party without being heard by the calling party on the call. It splits the calling party away so the Attendant can confidentially determine if the called party can accept the call.

Direct Trunk Group Selection

Allows the Attendant direct access to an idle outgoing trunk by pressing the button assigned to the trunk group. This feature eliminates the need for the Attendant to memorize, or look up, and dial the trunk access codes associated with frequently used trunk groups. Pressing a labelled button selects an idle trunk in the desired group.

Accessing the Attendant

Recall

Allows users to recall the Attendant when they are on a two-party call or on an Attendant Conference call held on the console. Single-line users press the Recall button or flash the switchhook to recall the Attendant. Multi-appearance users press the Conference or Transfer button to recall the Attendant and remain on the connection when either button is used.

Emergency Access to the Attendant

Provides for emergency calls to be placed to an Attendant. These calls can be placed automatically by the system or can be dialed by system users. Emergency access calls will generally receive priority handling by the Attendant.

Dial Access to Attendant

Allows you to reach an Attendant by dialing an access code. The Attendant can then extend the call to a trunk or to another telephone.

Individual Attendant Access

Allows you to call a specific attendant console. Each attendant console can be assigned an individual extension number.

Monitoring Calls

Trunk Identification By Attendant

Allows an Attendant or display-equipped telephone user to identify a specific trunk being used on a call. This capability is provided by assigning a Trunk ID button to the attendant console or telephone. This feature is particularly helpful for identifying a faulty trunk. That trunk can then be removed from service and the problem quickly corrected.

Crisis Alert

Visibly and audibly alerts Attendants when an emergency call is placed. The feature indicates from where an emergency call is made, which allows the Attendant to direct emergency-service response to the caller. Though often used in the hospitality industry, it can be set up to work with any standard attendant console.

Audible alerting sounds like an ambulance siren. Visual alerting consists of flashing of the crisis-alert button lamp and display of the caller name and extension. When crisis alerting is active, the console is placed in position-busy mode so that no other incoming calls interfere with the emergency call. The console can still originate calls. The Attendant must press the position-busy button to unbusy the console and the crisis-alert button to deactivate audible and visual alerting.

Trunk Group Access

Allows an Attendant to control trunk groups and prevents telephone users from directly accessing a controlled trunk group. This allows the Attendant to monitor the use of these trunk groups. By watching the lamps associated with the trunk groups, the Attendant can determine if the number of busy trunks in a specific trunk group has reached a preset warning level and if all trunks in a specific trunk group are busy. The Attendant can then handle other calls to these trunk groups accordingly.

Direct Extension Selection With Busy Lamp Field

Allows the Attendant to keep track of extension status — whether the extension is idle or busy — and to place or extend calls to extension numbers without having to dial the extension number. The Attendant can use this feature in two ways: using standard Direct Extension Selection access, or using enhanced Direct Extension Selection access.

Trunk Group Busy/Warning Indicators to Attendant

Provides the Attendant with a visual indication that the number of busy trunks in a group has reached an administered level. A visual indication is also provided when all trunks in a group are busy. This feature is particularly helpful to show the Attendant that the Attendant Control of Trunk Group Access feature needs to be invoked.

Room Status (Hospitality Industry)

Allows an Attendant to see whether a room is vacant or occupied and the housekeeping status of each room. This feature is only available when you have Enhanced Hospitality enabled for your system. This feature combines the property management capabilities of Check-In/Check-Out and Housekeeping Status but does not require that you have a Property Management System.

Attendant Direct Trunk Group Selection

Allows the Attendant direct access to an idle outgoing trunk by pressing the button assigned to the trunk group. This feature eliminates the need for the Attendant to memorize, or look up, and dial the trunk access codes associated with frequently used trunk groups. Pressing a labelled button selects an idle trunk in the desired group.

Centralized Attendant Service

Enables Attendant services in a private network to be concentrated at a central location. Each branch in a Centralized Attendant Service has its own listed directory number or other type of access from the public network. Incoming calls to the branch, as well as calls made by users directly to the Attendants, are routed to the centralized Attendants over Release Link Trunks.

Attendant with DCS

Control of Trunk Group Access

Allows an Attendant at any node in the Distributed Communication System (DCS) to take control of any outgoing trunk group at an adjacent node. This is helpful when an Attendant wants to prevent telephone users from calling out on a specific trunk group for any number of reasons, such as reserving a trunk group for incoming calls or for a very important outgoing call.

Direct Trunk Group Selection

Allows the Attendant direct access to an idle outgoing trunk in a local or remote trunk group by pressing the button assigned to that trunk group. This feature eliminates the need for the Attendant to memorize, or look up, and dial the trunk access codes associated with frequently used trunk groups. Direct Trunk Group Selection is intended to expedite the handling of an outgoing call by the Attendant.

Display

Shows call-related information that helps the Attendant to operate the console. Also shows personal service and message information. Information is shown on the alphanumeric display on the Attendant console. Attendants may select one of several available display message languages: English, French, Italian, or Spanish. In addition, your company may define one additional language for use by users and Attendants on their display.

Inter-PBX Attendant Calls

Allows Attendants for multiple branches to be concentrated at a main location. Incoming trunk calls to the branch, as well as Attendant-seeking voice-terminal calls, route over tie trunks to the main location.

DCS With Reroute

A sophisticated DCS rerouting capability for optimizing trunks. When you transfer out of your AUDIX voice messaging system, for example, DEFINITY ECS sets up a new path that optimizes system resources. Similar to the rerouting capabilities used with Q-SIG.

Enhanced DCS

Enhanced DCS adds features to the existing DCS capabilities and requires the use of Italian TGU/TGE tie trunks. Additional features include:

- Exchanging information to provide Class of Restriction (COR) checking between switches in the EDCS network
- Providing call-progress information for the Attendant
- Allowing Attendant intrusion between a main and a satellite PBX
- Allowing a main PBX to provide DID/CO intercept treatment rather than the satellite PBX

4 —Computer-Related Calling Features

Data Modules

Data modules connect DEFINITY ECS with other communications equipment, changing protocol, connections, and timing as necessary.

DEFINITY ECS supports the following types of data module:

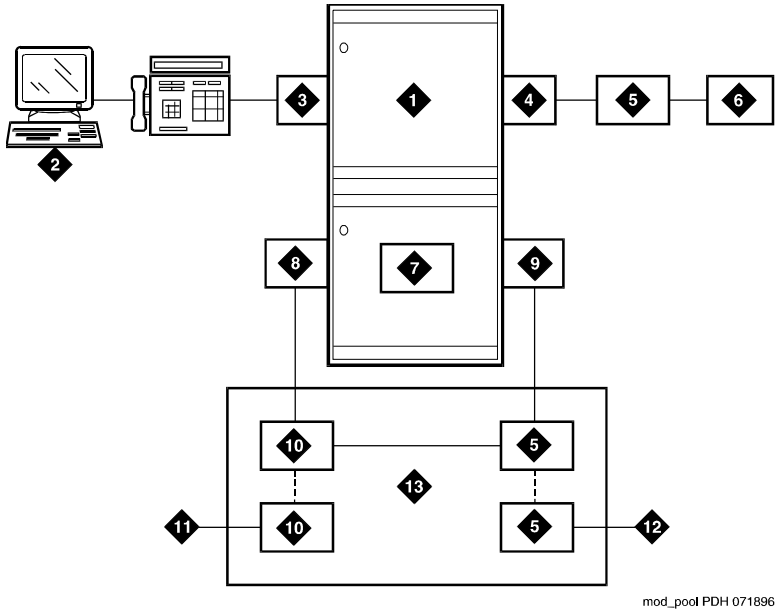
- High Speed Links
- Data stands
- Modular-processor data module
- 7000-series data modules
- Modular-trunk data module
- Asynchronous Data Unit
- Asynchronous Data Module (for ISDN-Basic Rate Interface telephones)
- Terminal adapters

All of these data modules support industry standards and include options for setting the operating profile to match that of the data equipment.

Modem Pooling

Enables switched connections between digital data endpoints (data modules) and analog data endpoints and acoustic coupled modems. Data transmission between a digital data endpoint and an analog endpoint requires a conversion since the DCP format used by the data module is not compatible with the modulated signals of an analog modem. A modem translates DCP format into modulated signals and vice versa. The Modem Pooling feature provides a set of modems for such conversions.

DEFINITY ECS modem pools are assigned into modem pool groups. A group can have up to 32 modems, called “members.” DEFINITY ECS can have as many as 63 modem pool groups. See Figure 9.



1	DEFINITY ECS	7	Integrated Pooled Modem
2	Asynchronous Terminal	8	Data Line Port
3	Digital Port	9	Analog Port
4	Analog Trunk	10	7400A
5	Modem	11	Digital Communications Protocol
6	Remote Application	12	Analog
		13	EIA Standard

Figure 9. DEFINITY ECS Modem Pooling

Alphanumeric Dialing

Allows you to place data calls by entering an alphanumeric name rather than a long string of numbers.

Data Call Setup

Enables the setting up of data calls using a variety of methods, such as: keyboard dialing, telephone dialing, Hayes command dialing, permanent switched connections, administered connections, automatic calling unit interface, and Hot Line dialing. Data Call Setup is provided for both DCP and ISDN-BRI telephones.

Data Hot Line

Provides for automatic placement of a data call when the originator hangs up. Data Hot Line may be used for security purposes. This feature offers fast and accurate call placement to commonly called data endpoints. Data terminal users who constantly call the same number can use Data Hot Line to automatically place the call when they hang up the telephone.

Default Dialing

Provides data terminal users who dial a specific number the majority of the time a very simple method of dialing that number. This feature enhances Data Terminal (Keyboard) Dialing by allowing a data terminal user to place a data call to a preadministered destination in several different ways, depending on the type of data module. Data Terminal Dialing and Alphanumeric Dialing are unaffected.

Data Privacy

Protects analog data calls from being disturbed by any of the system's overriding or ringing features. Data Privacy is activated when you dial an activation code at the beginning of the call.

Data Restriction

Protects analog data calls from being disturbed by any of the system's overriding or ringing features. It is administered at the system level to selected analog and multi-appearance telephones and trunk groups.

Administered Connections

Automatically establishes an end-to-end connection between two access or data endpoints based on administered attributes. This feature provides capabilities such as alarm notification, including an administrable alarm type and threshold; automatic restoration of connections established over a Software-Defined Data Network; ISDN-PRI trunk group [service may be referred to as ISDN-PRI (AC/AE) Service]; scheduled as well as continuous connections; and administrable-retry interval for failed connection attempts.

Multimedia Calling

Multimedia calls are initiated with voice and video only. Once a call is established, one of the parties may initiate an associated data conference to include all of the parties on the call who are capable of supporting data. The data conference is controlled by an adjunct device called an Expansion Services Module (ESM).

Multimedia Call Early Answer on Vectors and Stations

Early Answer is a feature applied to multimedia calls in conjunction with conversion to voice. Early Answer:

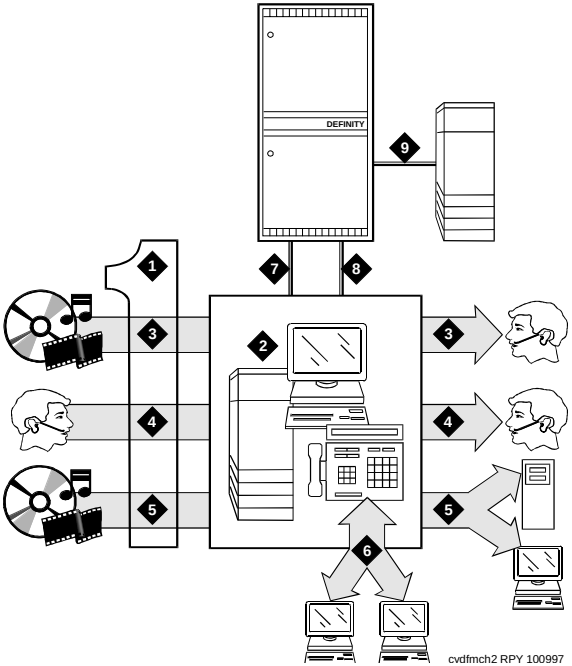
- Answers the data call
- Establishes the multimedia protocol prior to completion of a converted call
- Ensures that a voice path to/from the originator is available when the (voice) call is answered

For an incoming call, Early Answer answers the dynamic service-link calls when the destination endpoint answers, unless Early Answer is specified during routing or termination processing.

NOTE:
The “destination voice endpoint” might be an outgoing voice trunk if the destination voice station is forwarded or covered off-premises.

Multimedia Call Handling

Multimedia Call Handling (MMCH) enables you to control voice, video, and data transmissions using your telephone set. The feature buttons on a multi-function DEFINITY telephone enable you to conduct video conferences, and forward, cover, hold, or park multimedia calls much as you would a standard voice call. You can also share PC applications so that you and colleagues can collaborate while working from remote sites. See Figure 10.



1 One number access
2 Multimedia call complex
3 Multimedia to voice conversion
4 Standard voice call handling
5 Call redirection
6 Multimedia conferencing
7 BRI data connection
8 DCP voice connection
9 ESM data collaboration

Figure 10. DEFINITY ECS Multimedia Call Handling

Multimedia Call Redirection to MM Endpoint

A dual port multimedia station may be a destination of call redirection features such as call coverage, forwarding, and station hunting. The station can receive and accept full multimedia calls or data calls converted to multimedia.

Multimedia Data Conferencing (T.120) via ESM

The data conference is controlled by an adjunct device called an Expansion Services Module (ESM). The Expansion Services Module is used to terminate T.120 protocols [including Generalized Conference Call (GCC), a protocol standard for data conference control] and provide data conference control and data distribution. The MultiMedia Interface circuit pack, TN787, is used to rate adapt T.120 data to/from the ESM.

Multimedia Hold, Conference, Transfer, and Drop

Station users have the ability to activate hold, conference, transfer, or drop on multimedia calls. Multimedia endpoints and voice-only stations may participate in the same conference.

Multimedia Multiple-Port Network

DEFINITY ECS supports the equivalent of 580 Basic mode complexes operating at 6CCS traffic level. All enhanced mode complexes operate with soft-mode service links since the use of hard-mode service links reduces capacities. G3si limits are 1/3 to 1/2 of the G3r limits, depending on memory limitations and port network limitations.

Multiple Call Appearances

The dual port multimedia station provides multiple call appearances, each representing a multimedia call or a voice call.

Multimedia Applications Server Interface

The Multimedia Applications Server Interface provides a link between the DEFINITY ECS and one or more Multimedia Communications eXchange nodes. A Multimedia Communications eXchange is a stand-alone multimedia call processor produced by Lucent Technologies. This new link to DEFINITY ECS enhances the capabilities of each Multimedia Communications eXchange system by enabling it to share some of the DEFINITY ECS features. In particular, the interface provides the following advantages:

- Call Detail Recording (CDR)— The capture of call detail records so you can analyze the call patterns and usage of multimedia calls just as DEFINITY administrators analyze normal calls.
- Automatic Alternate Routing/Automatic Route Selection (AAR/ARS) — The intelligent selection of the most cost-effective routing for calls, based on available resources and your carrier preference. The system may select public trunks via DEFINITY Multimedia eXchange (MMCX)
- Voice Mail Integration — You can access your DEFINITY or INTUITY AUDIX voice messaging system from a Multimedia Communication eXchange (MMCX).

Local Area Network Connectivity (LAN)

The DEFINITY LAN Gateway feature lets the DEFINITY system exchange messages with PC/LAN-based communications software. The Control LAN (C-LAN) interface simplifies connections between adjunct equipment and DEFINITY. It provides TCP/IP connectivity over Ethernet or Point-to-Point Protocol connections to applications such as CMS Call Center, INTUITY AUDIX, and DCS.

IP Solutions

The capabilities and applications of DEFINITY ECS are extended with the introduction of IP Solutions. IP Solutions supports audio/voice over a LAN or WAN, and it ensures that remote workers have access to communication system features from their PCs.

administration

NOTE:

To maximize voice quality using IP Solutions, you must consider both your hardware and network configurations. For example, with IP Softphones, you can send the audio over traditional circuit switch lines, providing high quality voice, or over IP using LAN connections. When making calls over the LAN, voice quality will vary and may be unacceptable if you don't tune your computers and data network. For example, systems without full duplex audio cards and switch Ethernet ports are not recommended and will provide poor audio quality.

In addition, IP call capacities currently do not match those of traditional switch lines. See Chapter 4 in the *DEFINITY ECS R8.2 System Description* for more inofmration on call performance.

IP Solutions supports two trunk configurations and three types of softphones. It uses a new IP Interface circuit pack inside the DEFINITY ECS. This IP Interface is new with Release 8. It can operate in either of the following two modes:

- IP Trunk—used for IP trunk connections. This is the default mode for the Release 8 IP Interface, and this mode makes it compatible with Release 7 systems.

NOTE:

IP Trunk is the default mode for the Release 8 IP Interface only because it may also be used to replace the IP Interface in Release 7 systems. In a Release 8 system, however, the Release 8 IP Interface should be used in MedPro mode.

- MedPro—used for H.323 trunk connections and H.323 voice processing for IP softphones. This mode requires the new C-LAN circuit pack.

DEFINITY ECS supports multiple Release 8 IP Interfaces operating in a mixture of these two modes or a combination of Release 7 IP Interfaces (operating in the IP Trunk mode) and Release 8 IP Interfaces (operating in either mode). However, a Release 7 or Release 8 IP Interface operating in IP Trunk mode cannot communicate with a Release 8 IP Interface operating in MedPro mode.

Trunks

IP Solutions supports the following two types of trunks.

H.323

The Release 8 IP Interface in MedPro mode enables H.323 trunk service using IP connectivity between two DEFINITY ECS systems. The H.323 trunk groups can be

configured as DEFINITY-specific tie trunks, generic tie trunks, or direct-inward-dial (DID) "public" trunks. In addition, the H.323 trunks support ISDN features such as QSIG and BSR.

IP Trunk

The Release 8 IP Interface in IP Trunk mode allows for operation with existing Release 7 IP Interface circuit packs. It allows trunk groups to be defined as DS1 tie lines between DEFINITY ECS systems over a virtual private network.

IP SoftPhones

IP Solutions supports the following three SoftPhone applications.

Road-warrior Application

Enables use of the full DEFINITY ECS feature set from temporary remote locations anywhere in the world. The road-warrior application consists of two software applications running on a PC that is connected to DEFINITY ECS over an IP network. On DEFINITY ECS, the road-warrior application requires the C-LAN circuit pack for signaling and the Release 8 IP Interface running in MedPro mode for voice processing.

Telecommuter Application

Enables telecommuters to use the full DEFINITY ECS feature set from home. It consists of a PC and a telephone with separate connections to DEFINITY ECS. The PC provides the signaling path and the user interface for call control. A standard telephone provides a high-quality voice path. The telecommuter application requires the C-LAN circuit pack for signaling. The telecommuter application does not use the Release 8 IP Interface.

CentreVu IP Agent

Provides a variation of the telecommuter application. CentreVu IP Agent emulates a Lucent CallMaster set and provides use of the call end capabilities required for Call Center operations from a remote location, such as the agent’s home.

Integrated Workstation Applications

PassageWay

When you use the PassageWay adjunct application, you can use your Windows PC desktop to access the DEFINITY ECS telephone features. Users can access all available communications resources—telephone system, voicemail, fax, and LAN—from a single, graphical user interface.

CallVisor ASAI

The CallVisor Adjunct-Switch Application Interface (ASAI) lets external computer applications make use of the call routing, administration, and reporting capabilities of the DEFINITY ECS.

5 —Hospitality Features

Overview

The following features are designed for use in the hospitality industry. Other features listed elsewhere may be of use in this industry, however. The Attendant Crisis Alert feature is primarily used in lodging establishments. That feature is a basic feature because it is available on any system that has the appropriate attendant console.

Automatic Wakeup

Allows Attendants, front desk users, and guests to request that a wakeup call be placed automatically to a certain extension number at a later time. When a wakeup call is placed and answered, the system can provide a recorded announcement (which can be a speech synthesis announcement), music, or simply silence. With the Integrated Announcement feature, multiple announcements enable international guests to use wakeup announcements in a variety of languages.

Do Not Disturb

Allows guests, Attendants, and authorized front desk users to request that no calls, other than priority calls, be connected to a particular extension until a specified time.

Dual Wakeup

This capability is part of the Automatic Wakeup Hospitality Hotel/Motel feature. Dual Wakeup allows each extension to request two wakeup calls within one 24-hour period.

Room Activated Wakeup With Tones

This capability is part of the Automatic Wakeup Hospitality Hotel/Motel feature. Room Activated Wakeup With Tones allows guests to schedule wakeup calls via tones that prompt for the time they want to receive the wakeup call.

Hospitality Services

A system with Hospitality enabled and Hospitality Parameter Reduction disabled provides all system capabilities and supports all types of customers. A system with both Hospitality and Hospitality Parameter Reduction enabled provides reduced system parameters that have a major impact on essential system features used by nonlodging customers. The Hospitality features set (Auto Wakeup, Do Not Disturb, Property Management System) is the same on both packages.

Names Registration

Automatically sends a guest's name and room extension from the Property Management System (PMS) to the DEFINITY ECS at check-in, and automatically removes this information at check-out. The information may be displayed on any attendant console or display-equipped telephone at various hotel locations (for example, Room Service or Security).

Suite Check-In via the Hunt-to Feature

This feature is basically for the Hospitality Industry. Suite Check-In via the Hunt-to feature allows more than one station to be "checked- in" at one time. This is useful for a guest room that may have multiple extensions. This feature allows all extensions to be "checked-in" at the same time. Suite Check-In via the Hunt-to feature will also "check-out" all the extensions in the entire suite at the same time.

Property Management System Interface

Provides a communications link between the DEFINITY ECS and a Property Management System (PMS). The PMS allows a customer to control features used in both a hospital-type and a hotel/motel-type environment. The communications link allows the PMS to interrogate the DEFINITY ECS and allows information to be passed between the system and the Property Management System.

Property Management System (PMS) Digit to Insert/Delete

Many customer configurations base the room telephone extension on the room number by adding an extra leading digit. The PMS Insert/Delete Digit feature allows users to delete the leading digit of the extension in messages. The feature is useful for a hotel that has multiple extensions sharing an extra leading digit in front of the room number. The leading digit is automatically inserted when the message goes to the PBX.

NOTE:
The PMS interface supports 3-, 4-, or 5-digit extensions, but prefixed extensions do not send the entire number across the interface. Only the assigned extension number is sent. Therefore, you should not use prefixed extensions for numbers that are also going to use the Digit to Insert/Delete function.

Single-Digit Dialing and Mixed Station Numbering

Allows hotel staff and guests easy access to internal hotel/motel services and provides the capability to associate room numbers with guest room telephones. The feature provides the following dial plan types: single-digit dialing, prefixed extensions, and mixed numbering.

Auto-Selection of DID Numbers

Allows the hotel to give guests a telephone number that is not associated with the room number. The telephone number will be a Direct Inward Dialed (DID) number and will change each time someone checks in. There is an Automatic Digit Rotation that allows the number to change and be picked randomly from a block of DID numbers whenever a new guest checks in.

Station Hunt Before Coverage

This feature changes the interaction that occurs between station hunting and call coverage. Station Hunt before Coverage causes a call going to a busy station to go through a station hunting process before going to coverage. If all the stations in the Hunt group are busy, the call will go to the coverage path.

6 —System Management Features

Scheduling

DEFINITY ECS's functional scheduling allows you to specify the time a command will be executed or to specify that it should be executed on a periodic basis. Only commands that do not require user interaction after being entering on the command line (such as list, display, test) can be scheduled.

Concurrent User Sessions

In order to increase the efficiency of administration and maintenance functions, the DEFINITY ECS switch accommodates multiple concurrent administration and maintenance user sessions. Three or more devices (management terminals or operation support systems) can be connected to the switch to perform administration and/or maintenance tasks simultaneously. DEFINITY ECS supports eight concurrent administration and maintenance users — five can perform concurrent administration, and three can perform concurrent maintenance. The eight concurrent sessions can be in any combination of local and remote connections.

Security

Call Restrictions

By dialing an access code, Administrators and Attendants have the ability to restrict users from making or receiving certain types of calls. There are five restrictions:

- Outward — User cannot place external calls.
- Station-to-Station — User cannot place or receive internal calls.
- Termination — User cannot receive any calls (except priority calls).
- Toll — User cannot place toll calls but can place local calls.
- Total — User can neither place nor receive any calls.

The risks of unauthorized access can be minimized by combining the use of Remote Network Access with the following:

- An unpublished remote access number
- Deactivate unassigned barrier codes immediately
- Change barrier codes frequently
- Inform remote access users of their responsibility
- Monitor call detail reports for unauthorized or abnormal calling patterns

Class of Restriction (COR)

Defines many different classes of call origination and termination privileges. DEFINITY ECS may have no restrictions, only a single COR, or may have as many classes of restrictions as necessary to effect the desired restrictions. Many different types of classes of restriction can be assigned to many types of facilities on the switch. For example, you can use a calling-party COR to prevent callers from accessing the public network.

Security Violation Notification (SVN)

Security Violation Notification (SVN) allows you to set security-related parameters and to receive notification when the limits that you have established are violated. You can run reports related to both valid and invalid access attempts. You can also disable a login ID or remote access authorization that is associated with a security violation.

Station Security Codes

To provide additional security around the customer options the “init” login has been provided with additional security for the purpose of establishing an authentication procedure for attempts to remotely log into the system.

Calling/Connected Party Number (CPN) Restriction

Per Line CPN Restriction

Users may block the Calling Party Number when originating calls. For ISDN calls, the CPN Presentation Indicator is encoded accordingly. For non-ISDN calls, going to a public network that supports the CPN Restriction feature, the network specific Feature Activation Code gets passed to the network for interpretation and activation.

If Per Line CPN Restriction is administered for a station, it will override any ISDN Trunk Group administration for sending Calling Party Number.

Per Call CPN Restriction

Users may indicate Calling Number privacy information. For ISDN calls, the CPN Presentation Indicator is encoded accordingly. For non-ISDN calls going to a public network that supports the CPN Restriction feature, the network specific Feature Activation Code gets passed to the network for interpretation and activation of the desired feature.

If Per Call CPN Restriction is activated for an outgoing call, it will override any Per Line CPN Restriction administration for the calling station, and will override any ISDN Trunk Group administration for sending Calling Number.

Restriction — Controlled

Allows an Attendant or telephone user, with console permission, to activate and deactivate for an individual telephone or a group of telephones, the following restrictions: outward, total, station-to-station, and termination restrictions.

Malicious Call Trace

Allows you to trace malicious calls. You define a group of terminal users who can notify others in the group when they receive a malicious call. These users can then retrieve information related to the call. Using this information, you can identify the malicious call source or provide information to personnel at an adjacent system to complete the trace. It also allows you to record the malicious call.

CAMA - E911 Trunk Group

This form administers the Centralized Automatic Message Accounting (CAMA) trunks and provides Caller's Emergency Service Identification (CESID) information to the local community's Enhanced 911 system through the local Central Office.

Authorization Codes

Authorization Codes extend calling-privilege control and enhance security for remote-access callers.

Authorization codes may be used to:

- Override facility restriction levels assigned to originating stations or trunks
- Restrict individual incoming tie trunks and remote-access trunks from accessing outgoing trunks
- Track CDR calls for cost-allocation purposes
- Provide additional security control

Block Collect Call

Blocks collect calls. This feature is used primarily in Brazil.

Power Failure Transfer

Provides service to and from the local telephone company Central Office, including Wide Area Telecommunications System, during a power failure. This allows you to make or answer important or emergency calls during a power failure. This feature is also called Emergency Transfer.

DEFINITY Network Administration (DNA)

DEFINITY Network Administration (DNA) is a software-only system management tool for administering DEFINITY and voice mail systems. DNA is client-server based, runs on Windows, and is designed for large customers who have multiple DEFINITY systems. DNA allows multiple administrators to administer the same or different switches at the same time. It lets users schedule switch changes, save and modify switch tasks that run

regularly, and view a history of DNA activity. DNA allows for cut-thru to switches and voice mail systems, but it also offers the Graphically Enhanced DEFINITY Interface (GEDI) for DEFINITY systems. GEDI lets you do virtually everything you can do with a System Management Terminal (SMT), but using an easy-to-use graphical interface.

DNA also offers wizards that streamline and automate system management tasks, including adding or removing users; changing a user's name; adding hunt groups; adding, replacing, or deleting authorization codes; importing data into and exporting data from a switch; making global changes; printing button labels; viewing available extensions or ports; and setting the time on the switch. Optional features that can be purchased separately include Enhanced Number Portability, which streamlines station moves across networked switches.

DEFINITY Site Administration (DSA)

DEFINITY Site Administration (DSA) is another software-only system management tool for DEFINITY and voice mail systems. Designed for small to mid-sized customers, DSA offers everything that DNA offers except support for multiple, simultaneous administrators, and select wizards and DNA features.

Like DNA, DSA offers cut-thru, GEDI, task scheduling and editing, and history viewing. DSA wizards include adding and removing users; changing a user's name; adding bridged appearances; browsing for dial ranges, stations, and unused ports; finding unused extensions; importing data into and exporting data from a switch; making global changes; printing button labels; creating and using templates; printing reports; and setting the time on a switch. DSA also includes wizards for monitoring switch errors and performance, including monitor alarms, monitor trunks, trunk analyzer, processor occupancy, call traffic, system capacity, audits, and hardware manager.

Troubleshooting

Automatic Circuit Assurance

Assists in identifying possible trunk problems. The DEFINITY ECS maintains a record of the performance of individual trunks and automatically calls a designated user when a possible failure is detected. This feature provides better service through early detection of faulty trunks and consequently reduces out-of-service time.

DCS Automatic Circuit Assurance

Allows a user or Attendant at one node to activate or deactivate Automatic Circuit Assurance referral calls for the entire DCS network. This transparency allows the referral calls to originate at a node other than the node that detects the problem.

Busy Verification of Terminals and Trunks

Allows Attendants and users of multi-appearance telephones to make test calls to trunks, telephones, and hunt groups to check the status of an apparently busy resource. With this feature, an Attendant or multifunction telephone user can distinguish between a telephone that is truly busy and one that only appears busy because of some problem. You can also use the feature to quickly identify faulty trunks.

Facility Busy Indication

Allows users of multi-appearance telephones to see which lines, trunk groups, terminating extension groups, hunt groups, or paging zones (called resources or facilities) are busy. When the lamp associated with the resource is lit, the resource is busy.

You can store extension numbers, trunk group access codes, and Loudspeaker Paging access codes in a Facility Busy Indication button. The Facility Busy Indication button provides direct access to any of the facilities.

Facility Test Calls

Allows telephone users to make test calls to access specific trunks, dual tone multifrequency receivers, time slots, and system tones. The user dials an access code and makes the test call to make sure the facility is operating properly. Security measures are included to prevent unauthorized use.

Packet Error History

Provides a 24-hour history of important packet level statistics that indirectly indicate some LAN performance characteristics. The 24-hour history gives the ability to look back at these measures if the trouble cleared.

Variable Length Ping

Provides an enhancement to the ping command included in R7.1. This enhancement specifies a longer packet to be sent by ping and shows if a router or host has a problem fragmenting or integrating transferred packets.

Information and Reports

Recent Change History

Allows the system manager to view or print a history report of the most recent administration and maintenance changes on the switch. This report may be used for diagnostic or information purposes.

Service Observing

Allows a specified user, such as a supervisor, to observe or monitor another user's calls. A vector directory number call can also be observed. Observers can observe in listen-only or listen-and-talk mode. You set up Service Observing to observe a particular extension, not all calls to all extensions at a terminal.

NOTE:
Service Observing may be subject to federal, state, or local laws, rules, or regulations or require the consent of one or both of the call parties. Familiarize yourself and comply with all applicable laws, rules, and regulations before using this feature.

Calling Party/Billing Number (CPN/BN)

Allows the system to transmit Calling Party Number/Billing Number (CPN/BN) information to an ISDN-PRI trunk group. The CPN is the calling party's telephone number. BN is the calling party's billing number. The CPN/BN may contain international country codes. It is used with an adjunct application.

Call Charge Information

Provides two ways to know the approximate charge for calls made on outgoing trunks:

- Advice of Charge — For ISDN trunks

  Advice of Charge (AOC) collects charge information from the public network for each outgoing call. Charge advice is a number representing the cost of a call; it is recorded as either a charging or currency unit.

- Periodic Pulse Metering — For non-ISDN trunks

  Periodic Pulse Metering (PPM) accumulates pulses transmitted from the public network at periodic intervals during an outgoing trunk call. At the end of the call, the number of pulses collected is the basis for determining charges.

Call-charge information helps you to account for the cost of outgoing calls without waiting for the next bill from your network provider. This is especially important in countries where telephone bills are not itemized. You can also use this information to let employees know the cost of their phone calls, and so encourage them to help manage the company's telecommunications expenses.

NOTE:
This feature is not offered by the public network in some countries, including the United States.

In addition, the Pass Advice of Charge to BRI endpoints feature will transparently pass AOC information that has been received from PRI networks to WCBRI endpoints.

Bulletin Board

Provides a place on the switch where you can post information and receive messages from other switch users, including Lucent Technologies personnel. Anyone with appropriate permissions can use the bulletin board for everyday messages. In addition, Lucent Technologies personnel can leave high-priority messages, which are displayed on the first 10 lines of the bulletin board.

Call Detail Recording (CDR)

Records detailed call information on incoming and outgoing calls for the purpose of call accounting and sends this call information to a Call Detail Recording (CDR) output device. You can specify the trunk groups and extensions for which you want records to be kept as well as the type of information to be recorded. You can keep track of both internal and external calls. This application contains a wide variety of administrable options and capabilities.

Traffic Reports

Shows measurements in the form of switch-based reports for local or remote access, and can be collected for subsequent analysis and reporting by adjuncts and operation support systems using the operation support system interface protocol. These reports include:

- Call Coverage reports, which display measurements of the distribution of traffic offered to call-coverage groups. Separate reports for all calls and external calls are supplied.
- Coverage Points, which differs based on whether “All Calls” or “External Calls” is selected. For each coverage point in the group, the number of calls offered, abandoned while at that coverage point, and overflowing to the next coverage point are listed.
- Processor Occupancy report, which provides summary information on how heavily the processor is loaded.
- Traffic Summary report, which provides a performance summary of the DEFINITY ECS
- Attendant Position report, which lists the following:
  - Attendant usage
  - Number of calls answered
  - Total time the attendant was available to answer a new call
  - Average holding time on calls answered
- Security Violations report
- Tandem Traffic report, which provides information on facilities that serve tandem traffic
- Hunt Group Measurements

- Automatic Route Selection Pattern Measurements
- Trunk Group Detailed Measurements
- Blockage Study report
- Port Network and Link Usage
- Emergency and Journal reports coming from information from Crisis Alert to Digital Paging

Announcements

Recorded Announcement

Provides a recorded announcement to a variety of types of calls: calls that cannot be completed as dialed, calls that have been in queue for an assigned interval, any calls whose destination is an announcement, or incoming calls to a user.

Music-on-Hold Access

Automatically provides music, silence, or tone to a caller. Music lets the caller know that the connection is still valid. Many different music options can be administered to accommodate different tenants on the DEFINITY ECS. See the Tenant Partitioning feature for more information.

Multimedia Queuing with Voice Announcement

When multimedia callers queue for an available member of a hunt group they are able to hear an audio announcement.

Class of Service (COS)

Defines whether or not telephone users *can* access the following features and functions: Automatic Callback, Call Forwarding, Data Privacy, Priority Calling, Restrict Call Forwarding Off-Net, Call Forward Busy/Don't Answer, Personal Station Access, Extended Forwarding and Busy/Don't Answer, Trunk-to-Trunk Transfer Restriction Override, Off-Hook Alert, Console Permission, or Client Room.

See "Class of Restriction (COR)" on page 66.

Administration Without Hardware

Allows you to administer telephones that are not yet physically present on the system. This feature works the same as administration with hardware: when stations are moved, user-activated features such as Call Forwarding and Send All Calls are preserved and functional. This greatly facilitates the speed of setting up and making changes to the telephones on the system.

Terminal Translation Initialization (TTI)

DEFINITY ECS provides Terminal Translation Initialization, a feature that works with Administration Without Hardware. Terminal Translation Initialization associates the terminal translation data with a specific port location through the entry of a special feature-access code, a Terminal Translation Initialization security code, and an extension number from a terminal that is connected to a wired (but untranslated) jack.

Tenant Partitioning

Allows partitioning of the DEFINITY ECS in order to lease the system's services and features to tenants. This provides attractive new services and revenue for "virtual" landlords. It provides the robust features of a large system at affordable rates to small business tenants. DEFINITY ECS supports up to 100 partitions and 27 Attendant Groups. Multiple Attendant Groups can be assigned to each partition. Stations, hunt groups, and other endpoints assigned to a Class of Service can be partitioned. Network routing pattern preferences also support the assigned Tenant Partitioning. Tenant Partitioning also allows you to assign a unique music source for each tenant partition for customers who are put on hold.

See also, "Music-on-Hold Access" on page 74.

Trunk Management

DEFINITY ECS supports a variety of interfaces to voice and data networks. Trunks supply links between DEFINITY ECS, the public network, and other switches.

Trunk Group Circuits

Trunks provide the communications links between DEFINITY ECS and other switches, including Central Office switches and other premises switches. Trunks that perform the same function are grouped together and administered as trunk groups. Trunks interface with DEFINITY ECS via port circuit packs. DEFINITY ECS trunk group circuit types include the following:

Local Exchange Trunks

Local exchange trunks connect DEFINITY ECS to a Central Office. The following are some of the types available:

- Central Office trunks, which connect DEFINITY ECS to the local Central Office for incoming and outgoing calls
- Foreign Exchange trunks, which connect DEFINITY ECS to a Central Office other than the local one
- Wide Area Telecommunications Service (WATS) trunks, which allow you to place long-distance outgoing voice-grade calls to telephones in defined service areas. The calls are priced according to distance in the service area, length of the call, time of day, and the day of the week
- 800-service trunks, which let your business pay the charges for inbound long-distance calls so that callers can reach you toll-free
- Direct Inward Dialing (DID) trunks, which connect DEFINITY ECS to the local Central Office for incoming calls dialed directly to stations without Attendant assistance
- Digital Service 1 (DS1) trunks, which can be used to provide T1 or ISDN Primary Rate Interface (PRI) service

Tie Trunks

Tie trunks carry communications between DEFINITY ECS and other switches in a private network. Several types of trunks can be used, depending on the type of private network you establish.

Auxiliary Trunks

Auxiliary trunks connect devices in auxiliary cabinets with the DEFINITY ECS. Some of the features that are supported with this type of trunk are recorded announcements, telephone dictation service, malicious call trace, and loudspeaker paging.

Central Office: The location housing telephone switching equipment that provides local telephone service and access to toll facilities for long-distance calling.

Advanced Private Line Termination (APLT)

Provides access to and termination from CO (Central Office)-based private networks; namely, Common Control Switching Arrangements (CCSA) and Enhanced Private Switched Communications Service (EPSCS). APLT trunks are physically the same as those used for analog tie trunks, where the trunk signaling is compatible with EPSCS and CCSA network switches. The outgoing APLT trunk repeats any number of digits to the private network as dialed. APLT trunks can tandem through the PBX from EPSCS network only; CCSA networks require an Attendant to complete the call.

Direct Inward/Outward Dialing (DIOD)

Traditionally, CO (Central Office) trunks and DID (Direct Inward Dialing) trunks interface a PBX with a Central Office. A CO trunk services outgoing calls and accepts incoming calls that are terminated at the Attendant. A DID trunk is used for calls that need to be terminated without an Attendant interaction.

The DIOD trunk combines the features of a CO trunk and a DID trunk to provide both outgoing and incoming calls with addressing information in both directions.

IP Trunks

IP trunk groups may be defined as a virtual private network's tie lines between DEFINITY ECS systems or ITS-E servers. Each DEFINITY IP Trunk circuit pack provides a basic twelve-port package that can be expanded up to a total of 30 ports. The number of ports that are defined will correspond to the total number of simultaneous calls transmitted over the IP Trunk Interface.

The benefits of IP Trunk include a reduction in long distance voice and fax expenses, facilitating global communications, providing a full function network with data and voice convergence and optimizing networks by using the available network resources.

ISDN Trunks

Gives you access to a variety of public and private network services and facilities. The ISDN standard consists of layers 1, 2, and 3 of the Open System Interconnect (OSI) model. DEFINITY ECS can be connected to an ISDN using standard frame formats: Basic Rate Interface (BRI) and the Primary Rate Interface (PRI).

An ISDN provides end-to-end digital connectivity and uses a high-speed interface which provides service-independent access to switched services. Through internationally accepted standard interfaces, an ISDN provides circuit or packet-switched connectivity within a network and can link to other ISDN supported interfaces to provide national and international digital connectivity.

Integrated Services Digital Network—(Basic Rate Interface (ISDN-BRI)

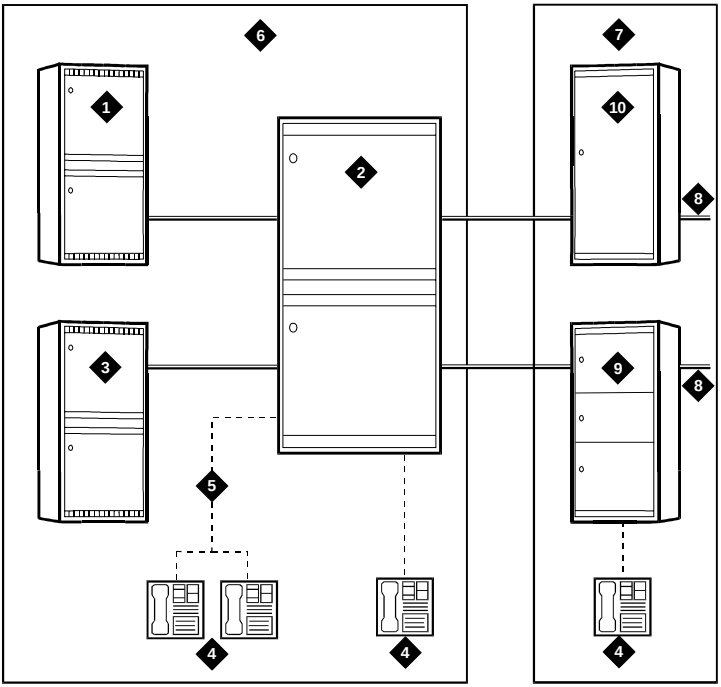
Enables connection of the system to equipment or endpoints that support an Integrated Services Digital Network (ISDN) by using a standard format called the Basic Rate Interface (BRI). This feature is a 192-Kbps interface that carries two 64-Kbps B-channels and one 16-Kbps D-channel.

ISDN is a global access standard that uses a layered protocol. It eliminates the need for multiple, separate access arrangements for voice, data, facsimile, and video services and networks. Using the same pair of wires that now carry simple telephone calls, ISDN can deliver voice, data, and video services in a digital format.

The ISDN-BRI Trunk circuit pack allows DEFINITY ECS to support the T interface and the S/T interface as defined by ISDN standards (ITU-T recommendation I.411). The circuit pack provides eight ports to the network and supports two B channels and one D channel. The ISDN-BRI Trunk provides the following advantages:

- Provides an inexpensive way to connect to ISDN services provided by the network provider
- Meets almost all ETSI Country protocol requirements
- Supports essential (not supplementary) ISDN services

BRI trunks support public-network access outside the U.S. on point-to-midpoint connections, with the restriction that DEFINITY ECS must not be configured in a passive bus arrangement with other BRI endpoints. ISDN-BRI trunks can be used as inter-PBX tie lines using the QSIG peer protocol.



1	DEFINITY ECS	6	Private ISDN
2	DEFINITY ECS	7	Public ISDN
3	DEFINITY ECS	8	Public and Private Networks
4	Basic Rate Interface Telephone	9	Central Office Switch
5	Passive Bus	10	Tandem Switch

Figure 11. DEFINITY ECS and ISDN

Full ETSI Functionality

The full set of ETSI public-network and private-network ISDN features is officially supported. This includes Look-Ahead Interflow, Look-Ahead Routing, and Usage Allocation. It also includes all QSIG supplementary services supported through R8:

- Name Identification
- Call Diversion (including rerouting)
- Call Transfer
- Path Replacement

It does not include:

- DCS
- Non-Facility Associated Signaling
- D-Channel Backup
- Wideband Signaling

NT Interface on TN556C

DEFINITY ECS supports the NT (network) side of the T interface using the TN556C circuit pack. This gives the switch full tie trunk capability using BRI trunks. DEFINITY ECS supports leased BRI connections through the public network, with a TN2185 on each end of the leased connection. DEFINITY ECS will not, however, allow customers to administer both endpoints and trunks on the same TN556C circuit pack.

Call-by-Call Service Selection

Enables a single ISDN-PRI trunk group to carry calls to a variety of services, rather than requiring each trunk group to be dedicated to a specific service. It allows you to set up various voice and data services and features for a particular call.

Facility and Non-Facility Associated Signaling

Allows an ISDN-PRI DS1/E1 interface D-channel to carry signaling information for B-channels (voice or data). D-Channel Backup can also be administered to increase system reliability.

Wideband Switching

Provides the ability to dedicate 2 or more ISDN B-channels or DSO endpoints for applications that require large bandwidth. Certain applications, such as video conferencing and high-speed data transmission, require extra bandwidth and it becomes

necessary to put several ISDN-PRI narrowband channels into one wideband channel to accommodate the needs of these applications. This feature supports both European and North American standards.

Multiple Subscriber Number (MSN) - Limited

The ISDN standard MSN feature lets customers assign multiple extension to a single BRI endpoint. The MSN feature works with BRI endpoints that allow the Channel ID IE to be encoded as “preferred.”

Automatic TEI (Termination Endpoint Identifier)

The user side will support automatic TEI assignment by the network. Both fixed and automatic TEI assignment will be supported on the network side.

ATM-CES Trunks

ATM-CES (Circuit-Emulation Service) lets the DEFINITY ECS emulate an ISDN-PRI trunk on an ATM facility.

ATM Trunks

Support telephony and Wide-Area Network (WAN) connectivity over ATM networks.

Personal Central Office Line (PCOL)

Provides a dedicated trunk circuit between multi-appearance voice terminals and a CO or other switch via the network.

ISDN Presentation Restriction

Replaces "Presentation Restricted" or "Unavailable" Calling/Connected number with an administered text for display. The administration options can be controlled on a per-ISDN-Trunk-Group basis.

ISDN Feature Plus

Enables those without DID to direct dial users on a remote PBX via the public network. ISDN Feature Plus eliminates the need for Attendant intervention for those without DID capabilities.

Tandem

In an Electronic Tandem Network, DEFINITY ECS provides a variety of features on a network-wide basis. Here are a few examples:

- Uniform Dial Plan (UDP) — A unique four- or five-digit number assigned to each station on the network. Uniform numbering gives each station a unique number (location code plus extension) that can be used at any location in the Electronic Tandem Network to access that station, DEFINITY ECS enhances the standard UDP with the unrestricted 5-digit Uniform Dial Plan, which allows up to five digits to be parsed for call routing.
- Extension Number Portability — When employees move within the network, they can retain their extension numbers. The ability to keep extension numbers, and even Electronic Tandem Network and Direct Inward Dialed numbers, when moving to other locations within the company eliminates missed calls and saves valuable time.
- Traveling Class Marks — Traveling Class Marks are a mechanism for passing a caller's facility restriction level from one Extended Tandem Network switch to another. Traveling Class Marks allow privilege checking to be passed across switches through the Electronic Tandem Network.
- Automatic Alternate Conditional Routing — You can control the routing of particular calls using conditional routing. For example, you can limit the number of communications satellite hops (communications satellite links used as trunks) in any end-to-end private network routing pattern. Limiting the number of satellite hops may be desirable for controlling transmission quality or call delay in both voice and data calls.
- Enhanced Trunk Signaling and Error Recovery — The reliability of Electronic Tandem Network calls is improved by allowing a trunk call to be retried on another circuit when signaling failures occur.

tandem switch: A switch within an electronic tandem network (ETN) that provides the logic to determine the best route for a network call, possibly modifies the digits outpulsed, and allows or denies certain calls to certain users.

tandem through: The switched connection of an incoming trunk to an outgoing trunk without human intervention.

tandem tie-trunk network (TTTN): A private network that interconnects several customer switching systems.

See also, "ATM-Port Network Connectivity (ATM-PNC)" on page 107.

Digital Multiplexed Interface

Supports two signaling techniques: bit-oriented signaling and message-oriented signaling for direct connection to host computers.

Digital Multiplexed Interface offers two major advantages. It delivers a standard, single-port interface for linking host computers internally and externally via T1 carrier. And, since it is compatible with ISDN standards and is licensed to numerous equipment manufacturers, it promotes multi-vendor connectivity.

DEFINITY ECS supports two versions of Digital Multiplexed Interface, each differing in the way information is carried over the 24th channel:

- Digital Multiplexed Interface-Bit-Oriented Signalling carries framing and alarm data and signalling information for connections to host computers and other vendor equipment.
- Digital Multiplexed Interface Message-Oriented Signalling, fully compatible with ISDN-PRI, uses the same message-oriented signalling format, Link Access Procedure on the D-channel, as ISDN-PRI for control and signalling. These signalling capabilities extend the advantages of Digital Multiplexed Interface-Message Oriented Signalling multiplexed communications to the public ISDN network.

Miscellaneous Trunks

Miscellaneous trunks perform functions that do not fit neatly into any of those already described:

- Release-link trunks are used between switch locations to provide Centralized Attendant Service or Automatic Call Distribution group availability.
- Remote-access trunks provide off-premises users with access to DEFINITY ECS features and networking.

Digital Interfaces

E1 Interface

DEFINITY ECS also supports E1 connections. T1/E1 access and conversion allows simultaneous connection to both T1 (1.544 Mbps) and E1 (2.048 Mbps) facilities (using separate circuit packs).

T1 Interfaces

When planning your networking requirements, one of the options you should consider is multiplexing over Digital Services 1 (DS1) facilities.

Used to connect switches to the public network or to other switches in a private network, DS1 also delivers high-speed, end-to-end digital connectivity. Voice and data calls are completed at transmission speeds of up to 64 kbps.

DS1 Trunk Service

Bit-oriented signaling that multiplexes 24 channels into a single 1.544-Mbps stream. DS1 can be used for voice or voice-grade data and for data-transmission protocols. E1 trunk service is bit-oriented signaling that multiplexes 32 channels into a single 2.048-Mbps stream. Both DS1 and E1 provide a digital interface for trunk groups.

Answer Detection

For purposes of Call-Detail Recording (CDR), it is important to know when the called party answers a call. DEFINITY ECS provides three ways to determine whether the far end has answered an outgoing call.

- Answer Detection — A call-classifier board detects tones and voice-frequency signals on the line and determines whether a call has been answered. This method is fairly accurate.
- Network Answer Supervision- The Central Office (CO) sends back a signal to indicate that the far end has answered. If a call has traveled over a private network before reaching the CO, the signal is transmitted back over the private network to the originating system. This method is extremely accurate, but is not available in the United States over CO, FX, or WATS trunks.
- Answer Supervision by Time-out — You set a timer for each trunk group. If the caller is off-hook when the timer expires, the DEFINITY ECS assumes that the call has been answered. This is the least accurate method. Calls that are shorter than the timer duration do not generate call records, and calls that ring for a long time produce call records whether they are answered or not.

Automatic Transmission Measurement System

Measures voice and data trunk facilities for satisfactory transmission performance. The measurement report contains data on trunk signal loss, noise, signaling return loss, and echo return loss. Acceptable performance, the scheduling of tests, and report contents are administrable.

Automatic Routing Features

DEFINITY ECS provides a variety of automatic-routing features for public and private networks. Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) are the foundation for these automatic-routing features. They route calls based on the preferred (normally the least expensive) route available at the time the call is placed. Generally, AAR routes calls over a private network and ARS routes calls using the public network numbering plan. However, both AAR and ARS support public and private networks. You can use the other features listed in this section when you use AAR and ARS.

Automatic Alternate Routing (AAR)

Allows private network calls to originate and terminate at one or many locations without accessing the public network. When you dial an access code and phone number, AAR selects the most desirable route for the call and performs digit conversion as necessary. If the first choice route is unavailable, another route is chosen automatically.

The numbers you call using AAR are normally private-network numbers. However, you can call a public-network number, a service code, an international number, operator access code, or an operator-assisted dialing number. With AAR and Subnet Trunking, you have a convenient way to place international calls to frequently-called foreign cities. Such calls route as far as possible over the private network, and then access the public network. This saves toll charges and allows you to use your private network as much as possible.

Automatic Route Selection (ARS)

ARS selects carriers automatically and routes calls inexpensively over the public network. When there are one or more long-distance carriers and Wide-Area Telecommunications Services (WATS) provided, DEFINITY ECS selects the most preferred route for the call. Long-distance carrier-code dialing is not required on routes selected by the system. You

assign long-distance carrier-codes and DEFINITY ECS translates them. The system inserts codes as needed to guarantee automatic-carrier selection. ARS can route calls to a variety of types-of-numbers and access a variety of types of trunk groups.

AAR/ARS Overlap Sending

DEFINITY ECS supports overlap sending for AAR and ARS calls that are routed over ISDN-PRI trunk groups. ISDN-PRI call-address information is sent one digit at a time instead of in one block. In countries with complex public-network numbering plans, this allows for a significant decrease in call setup time. When overlap receiving is enabled, this is especially significant for tandemed calls.

AAR/ARS Partitioning

Allows AAR and ARS to be partitioned into 8 user groups within a single DEFINITY ECS and provides individual routing treatment for each of these user groups.

User groups share the same Partition Group Number, which indicates the choice of routing tables that are used on a particular call. Each Class of Restriction (COR) is assigned a specific Partition Group Number or Time of Day specification. Different classes of restriction may be assigned the same Partition Group Number.

Time of Day Routing

Provides the most economical routing of ARS and AAR calls. This routing is based on the time of day and day of the week that each call is made. Up to 8 TOD routing plans may be administered, each scheduled to change up to 6 times a day for each day in the week.

This allows you to take advantage of lower calling rates during specific times of the day and week. In addition, companies with locations in different time zones can use different locations that have lower rates at different times of the day or week. This feature is also used to change patterns during the times an office is closed in order to reduce or eliminate unauthorized calls.

Facility Restriction Levels and Traveling Class Marks

Allows certain calls to specific users, while denying the same calls to other users. For example, certain users may be allowed to use Central Office trunks to other corporate locations while other users may be restricted to less expensive private-network lines. You can administer up to eight levels of restriction for users of AAR and ARS.

Alternate Facility Restriction Levels

Allows DEFINITY ECS to adjust facility restriction levels or authorization codes for lines or trunks. Each line or trunk is normally assigned a facility restriction level. With this feature, Alternate Facility Restriction Levels are also assigned. Attendants can change to the alternates, thus changing access to lines and trunks. You might want to use this feature to disable most long-distance calling at night, for example, to prevent unauthorized staff from making long-distance calls.

CAUTION:
This feature may change the AAR and ARS routing preferences. Using it on tandem and tie-trunk applications affects entire networks. Calls that are part of a cross-country private network may be blocked.

Generalized Route Selection

Provides voice and data call-routing capabilities. You use it to select not only the least-cost routing, but also optimal routing over the appropriate facilities. It enhances AAR and ARS by providing additional parameters in the routing decision and maximizing the chance of using the right facility to route the call. Also, if an endpoint incompatibility exists, it provides a conversion resource (such as a modem from a modem pool) to attempt to match the right facility with the right endpoint.

Look-Ahead Routing

Provides an efficient way to use trunking facilities. It allows you to continue to try to reroute an outgoing ISDN-PRI call that is not completing. When DEFINITY ECS receives a cause value that indicates congestion, Look-Ahead Routing tells the system what to do next. For each routing preference, you can indicate if the next routing-preference should be attempted or if the current routing-preference should be attempted again.

Subnet Trunking

Modifies the number you dial so an AAR or ARS call can route over different trunk groups that may terminate in switches with different dial plans. Subnet Trunking inserts digits, deletes digits, pauses, and/or waits for dial tone in digit outpulsing, as required, so calls route:

- To or through a remote switch
- Over Tie trunks to a private network switch
- Over CO trunks to the serving CO

Subnet Trunking is required on calls routing to or through a remote switch, regardless of the call's destination.

Extension Number Portability

Gives you the ability to assign any extension to any system in a subnetwork. Stations can be moved across systems while retaining the original extension number, as long as the systems are part of a defined subnetwork. This feature is used in conjunction with Automatic Alternate Routing and Uniform Dial Plan.

Alarm Assignments

Alternate Operations Support System Alarm Number

Allows you to establish a second number for the DEFINITY ECS to call when an alarmable event occurs. This feature is useful for alerting a second support organization, such as INADS or OneVision.

External Device Alarming

Allows you to assign analog ports to alarm interfaces for external devices. You can specify a port location, information to identify the external device, and the alarm level to report when a contact closure occurs.

Customer-Provided Equipment Alarm

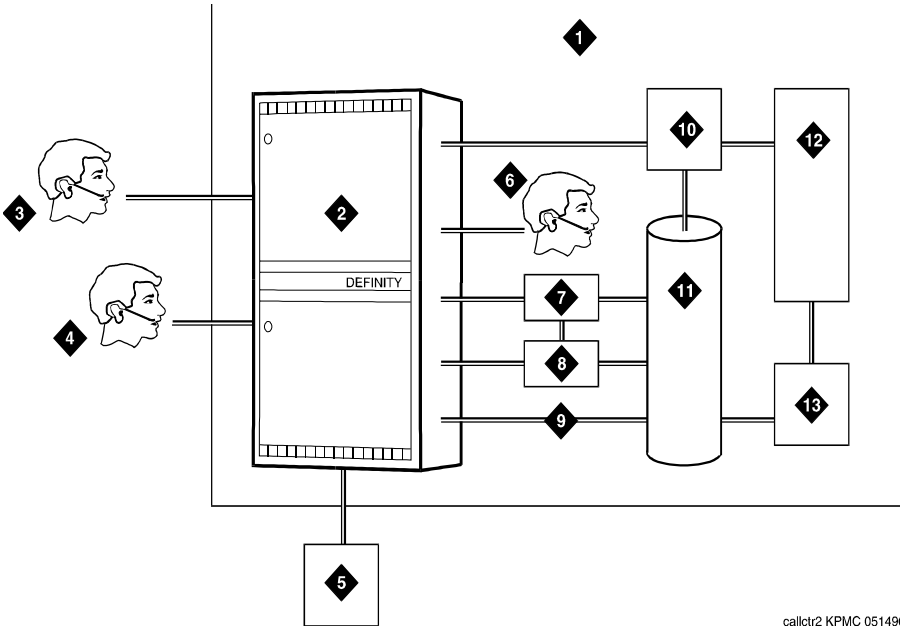
Provides you with an indication that a system alarm has occurred and that the DEFINITY ECS has attempted to contact a service organization. A device that you provide, such a lamp or a bell, is used to indicate the alarm situation. You can administer the level of alarm about which you want to be notified.

7 —Call Center Features

DEFINITY Call Center applications are designed to efficiently connect each caller with the representative best suited to serve that caller. The DEFINITY ECS begins the process by capturing information about the caller even before the call is routed. That information is integrated with existing databases and the combined data is used to match caller to agent. Additional DEFINITY features politely keep callers waiting in queue (a holding place for incoming calls) informed about how long it will probably take to process the call. Detailed call statistics are constantly available to agents and supervisors.

Calls coming into your DEFINITY ECS Call Center are queued up and routed based on information that the system continually acquires. Each of your customers can be presented with a variety of options for leaving a voice message, leaving a fax, or monitoring the status of his or her call. Using CONVERSANT voice response software, the system can even respond appropriately to spoken information.

Figure 12 summarizes how you might set up a DEFINITY Call Center.



1	Your Office Building	8	CentreVu Supervisor
2	DEFINITY ECS	9	CallVisor Adjunct Switch Applications Interface
3	Remote Agents Using DEFINITY Extender	10	INTUITY Conversant Voice Response System
4	Remote Agents (Home Agent)	11	Local Area Network
5	Remote Call Center Using Look Ahead Interflow or Best Service Routing	12	Host Computer
6	Local Agents Organized By Skill	13	Computer-Telephone Interface Server
7	CentreVu CMS		

Figure 12. A DEFINITY Call Center

Automatic Call Distribution (ACD)

ACD is the basic building block for Call Center applications. ACD offers you a method for distributing incoming calls efficiently and equitably among available agents. With ACD, incoming calls can be directed to the first idle or most idle agent within a group of agents.

Agents in an ACD environment are assigned to a hunt group, a group of agents handling the same types of calls. A hunt group is also known as a split or skill with EAS.

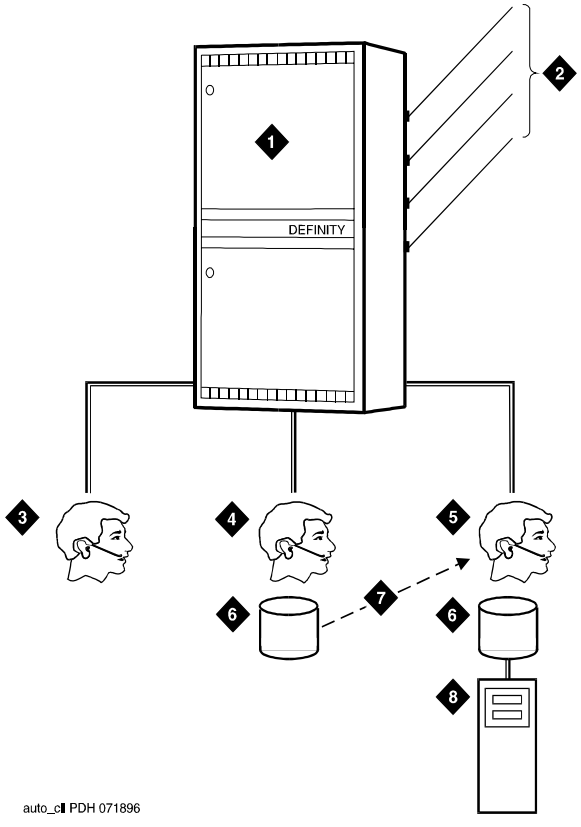
A hunt group is especially useful when you expect a high number of calls to a particular phone number. A hunt group might consist of people trained to handle calls on specific topics. For example, the group might be:

- A benefits department within your company
- A service department for products you sell
- A travel reservations service
- A pool of attendants

In addition, a hunt group might consist of a group of shared telecommunications facilities. For example, the group might be:

- A modem pool
- A group of data-line circuit ports
- A group of data modules

In the Figure 13 example, Hunt Group A receives calls only when agents are available since it has no queue. Calls to Hunt Group B can be queued while agents are unavailable, and redirected to Hunt Group C if not answered within an administrable time. Calls to Hunt Group C are redirected to voice mail if not answered within an administrable time.



1	DEFINITY ECS	5	Group C: General Information
2	Incoming Lines	6	Queues
3	Group A: Business Travel	7	Call Coverage to Group C
4	Group B: Personal Travel	8	Voice Mail

Figure 13. A Basic Example of Automatic Call Distribution

Agent Call Handling

Allows you to administer functions that ACD agents use when handling incoming calls. You define specific agent capabilities and can plan capacities based on those capabilities. The same list of agent capabilities are also supported through the CallVisor Adjunct/Switch Applications Interface (ASAI).

Multiple Call Handling

Allows agents to receive an ACD call while other types of calls are alerting, active, or on hold.

Auto-Available Split (AAS)

Allows members of an ACD split to be in Auto-In work mode continuously. An agent in Auto-In work mode becomes available for another ACD call immediately after disconnecting from an ACD call. You can use AAS to bring ACD-split members back into Auto-In work mode after a system restart. Although not restricted to such, this feature is intended to be used for splits containing only recorders or voice-response units.

Queue Status Indications

Allows you to assign Queue-Status Indicators for ACD calls based on the number of calls queued and time in queue. You can assign these indications to lamps on agent, supervisor, or attendant terminals or consoles to help monitor queue activity. In addition, you can define auxiliary queue warning lamps to track queue status. On display telephones, you can display the number of calls queued and time in queue of a split's oldest call.

Circular Station Hunting

This feature will eliminate the "hot seat" in a hunt group. The DEFINITY ECS will keep track of the last extension in the hunt group that has received a call. When another incoming call arrives, the next idle extension will receive the call, bypassing the extension that had received the previous call. The first extension in the hunt group will no longer be the busiest telephone while the others in the group are sitting idle.

Reason Codes

Allows agents to enter a numeric code that describes their reason for entering Auxiliary (AUX) work mode or for logging out of the system. Reason codes give Call Center managers detailed information about how agents spend their time. You can use this data

to develop more precise staffing forecasting models or use it with schedule-adherence packages to ensure that agents are performing scheduled activities at the scheduled time. You must have Expert Agent Selection (EAS) enabled to use reason codes.

Call Center Release Control

Determines which features are "active" on your switch. Call Center Release Control will control whether certain new Call Center software features are available to you.

Redirection on No Answer

Redirects a ringing ACD split or skill call or Direct Agent Call after an administered number of rings. This prevents an unanswered call from ringing indefinitely. The call can redirect either to the split or skill to be answered by another agent or to a Vector Directory Number (VDN) for alternative call handling. Direct Agent Calls route to the agent's coverage path, or to a VDN if no coverage path is administered. You must have ACD enabled to use this feature.

Intraflow and Interflow

Intraflow and Interflow allow you to redirect ACD calls from one split to another split. Intraflow redirects calls to other splits within the system using Call Coverage or Call Forwarding All Calls. Interflow redirects calls to an external split or location using Call Forwarding All Calls. You can have calls redirected from one split to another *conditionally*, according to the coverage path's redirection criteria. For example, you can define a split's coverage path to automatically redirect incoming ACD calls to another split when a terminal is busy or unanswered.

Look-Ahead Interflow

Balances the load of ACD calls across multiple locations. With Look-Ahead Interflow (LAI), you can optionally route a call to a backup location based on your system's ability to handle the call within parameters defined in a vector. In turn, the backup system can accept or deny the call also based on defined parameters.

Enhanced LAI allows interflowing only the call(s) at or near the head of the queue to provide First In/First Out (FIFO) or FIFO-like call distribution and significantly reduce call and trunk processing for LAI.

Enhanced information forwarding allows Call Center related information to be passed transparently over some public networks and non-QSIG or QSIG private networks using codeset 0 shared user-to-user information (UUI) (for non-QSIG) or QSIG Manufacturer-Specific Information (MSI). For more information about UUI, see "User-to-User Information Over the Public Network" on page 100.

Abandoned Call Search

Allows a Central Office that does not provide timely disconnect supervision to identify abandoned calls. An abandoned call is one in which the calling party hangs up before the call is answered. Abandoned Call Search is suitable only for older Central Offices that do not provide timely disconnect supervision.

Call Vectoring

Call Vectoring is a versatile method of routing incoming calls that can be combined with Automatic Call Distribution for maximum benefit and Call Center efficiency. A Call Vector is a series of call-processing steps (such as providing ringing tones, busy tones, music, announcements, and queuing the call to an ACD hunt group) that define how calls are handled and routed. The steps, called Vector Commands, determine the type of processing that specific calls will receive.

Vector commands may direct calls to on-premises or off-premises destinations, to any skill or hunt group, or to a specific call treatment such as an announcement, forced disconnect, forced busy, or music.

With combinations of different vector commands, incoming callers can be treated differently depending on the time or day of the call, the Expected Wait Time (EWT), the importance of the call, or other criteria. A DEFINITY ECS G3r can route incoming callers using up to 999 different vectors. A DEFINITY ECS G3i can route up to 256 and a G3si can route up to 48 different vectors. Each vector can have up to 32 commands. DEFINITY ECS also allows vectors to be linked via the "goto vector" command.

Vector Directory Numbers (VDN)

Calls access DEFINITY ECS vectors using Vector Directory Numbers (VDN). A VDN is a "soft" extension number that is not assigned to a physical equipment location. A Vector Directory Number has several properties that are administered by the System Manager.

A Vector Directory Number can be accessed in almost any way that an extension can be accessed.

When answering a call, the answering agent will see the information (such as the name) associated with the VDN on their display and can respond to the call with knowledge of the dialed number. This operation provides Dialed-Number Identification Service (DNIS), allowing the agent to identify the purpose of the incoming call.

COR for VDN

Class of Restriction is checked for transfer to the VDN. It can also be used to block the AUX Trunk announcement from some Agents. Observing can also be set to allow or restrict to that VDN.

VDN in a Coverage Path

VDN in a Coverage Path enhances Call Coverage and Call Vectoring to allow you to assign Vector Directory Numbers as the last point in coverage paths. Calls that go to coverage can be processed by vectoring/prompting to extend Call Coverage treatments.

VDN of Origin Announcement

VDN of Origin Announcement provides agents with a short message about a caller's city of origin or requested service based on the VDN used to process the call. VOA messages help agents respond appropriately to callers. For example, if you have two 800 numbers, one for placing orders and one for technical support, you can administer two VDNs to route calls to the same set of agents. When an incoming call is routed to a VDN with a VOA assigned (for example, "new order" or "tech help"), the VDN routes the call to a vector, which can place the call in an agent queue. When an agent answers the call, he or she hears the VOA message and can respond appropriately to the caller's request. This feature is particularly useful for visually impaired agents or agents that don't have display sets.

Applications

There are many different applications for Call Vectoring. However, Call Vectoring is used primarily to handle the call activity of Automatic Call Distribution hunt groups. Call Vectoring can also manage a queue by keeping calls queued in up to three hunt groups (with four different priority levels) while also providing a series of other processing options. Other common applications include:

- Special Treatment for Selected Callers
- Night Treatment
- Off-loading of Periodic Excess Calls
- Information Announcements for the Calling Party

Call Prompting

Allows the system to collect information from the calling party and direct the calls via Call Vectoring. The caller is verbally prompted by the system and enters information in response to the prompts. This information is then used to redirect the call or handle the call in some other way (taking a message, for example). This feature is mostly used to enhance the efficient handling of calls in the Automatic Call Distribution application.

Four applications are described below.

- Automated attendant — Allows the calling party to enter the number of any extension on the system. The call is then routed to the extension. This allows you to reduce cost by reducing the need for live attendants.
- DIVA (data in/voice answer) — Allows the calling party to hear selected announcements based on the digits that he or she enters. This may be used for applications such as an audio bulletin board.
- Data collection — Allows the calling party to enter data that can then be used by a host computer application to assist in call handling. For example, this data may be the calling party's account number, which could be used to support an inquiry/response application.
- Call Center messaging — Gives the calling party the option of leaving a message or waiting in queue for an agent. This may be used for an online order entry system or to further automate an incoming-Call Center operation.

Expert Agent Selection

Enables certain Expert Agent Selection skill types to be assigned to a call type or a Vector Directory Number. Routing calls via vectoring then allows the system administration to direct calls to agents who have the particular agent skills required to complete the customers' inquiries successfully.

Call Distribution Based on Skill

Calls that require certain agent skills (such as "speaks Spanish" or "knowledgeable about Product X") can be matched to an agent who matches the required skill. You can assign one of up to 999 skill numbers to each need or group of needs. The skills are administered and associated for each of the following:

- Vector Directory Numbers
- Agent Login IDs
- Callers

This refined skill definition capability allows you to organize call handling based on customer, product, and language, for example.

Add/Remove Skills

Allows an agent using Expert Agent Selection to add or remove skills. A skill is a numeric identifier that refers to an agent's specific ability. For example, an agent who speaks English and Spanish could be assigned a language -speaking skill with an identifier of 20. The agent then adds skill 20 to his or her set of working skills. If a customer needs a Spanish-speaking agent, the system routes the call to an agent with that skill. Each agent can have up to four active skills, and each skill is assigned a priority level.

Best Service Routing

Best Service Routing (BSR) distributes the call to the best local or remote split/skill among the resources to be considered, based on Expected Wait Time (EWT) or available agent characteristics.

Queue to Best ISDN Support

Queue to Best information is passed transparently over several public networks and QSIG private networks using the envelopes that are part of the QSIG Manufacturer-Specific Information (MSI) and the ISDN platform enhancement.

Dialed Number Identification Service (DNIS)

Displays, for a called party or answering position, the service or product associated with an incoming call. You administer what the system displays.

User-to-User Information Over the Public Network

Provides the mechanism to pass information across several key public networks, including information that is originated or destined for one of several applications on DEFINITY ECS.

Basic Call Management System (BCMS)

The Basic Call Management System helps you fine tune your Call Center operation by providing reports with the data necessary to measure your Call Center agents' performances.

This feature offers call management control and reporting at a low cost for Call Centers of up to 2000 agents. The BCMS collects and processes DEFINITY ECS's ACD call data (up to seven days) within the system; an adjunct processor is not required to produce Call Management reports.

The following are the types of reports that can be generated:

- Real-time reports
  - Agent Status
  - System Status
  - Vector Directory Number Status
- Historical reports
  - Agent
  - Agent Summary
  - Split
  - Split Summary
  - Trunk Group
  - Vector Directory Number report

CentreVu Call Management System (CentreVu CMS)

The CentreVu Call Management System collects call traffic data, formats management reports, and provides an administration interface for Automatic Call Distribution (ACD) on your DEFINITY ECS. It helps you manage the people, traffic load, and equipment in an ACD environment by answering such questions as:

- How many calls are we handling?
- How many callers abandon their calls before talking with an agent?
- Are all agents handling a fair share of the calling load?
- Are our lines busy often enough to warrant adding additional ones?
- How has traffic changed in a given ACD hunt group over the past year?

Dual Links to CMS

Provides an additional TCP/IP link to a separate CMS for full, duplicated CMS data collection functionality and High availability CMS configuration. The same data is sent to both servers and the administration can be done from either server. The ACD data will be delivered over different network routes to prevent any data loss from such conditions as ACD link failures, CMS hardware or software failures, CMS maintenance or CMS upgrades.

CMS Measurement of ATM

Provides the capability to externally measure ATM trunks on CMS. The CMS messages and reports are modified to support the expanded equipment location.

VuStats

VuStats presents Basic Call Management System (BCMS) statistics on telephone displays. Agents, supervisors, Call Center managers, and other users can press a button and view statistics for agents, splits or skills, VDNs, and trunk groups. These statistics can help agents monitor their own performance or respond appropriately to the caller's request.

Site Statistics for Remote Port Networks

Forwards location IDs to CMS to provide Call Center site-specific reports.

CentreVu Advocate

CentreVu Advocate is the collection of ECS features that provide new flexibility in the way a call is selected for an agent in a call surplus situation and in the way an agent is selected for a call. Instead of the traditional "First-In, First-Out" approach, The caller's needs, potential business value and their desire to wait are looked at and then the system will decide which agents should be matched to the callers.

Call Selection Override per Skill

Call Selection override is determined by skill. Call Center Supervisors can override the normal call handling activity on particular skills only, or for the entire Call Center.

Reserve Agent Time in Queue Activation

This feature activates a reserve agent if a skill's expected wait time (EWT) exceeds a pre-determined threshold or if the call's time in the queue exceeds the administered Service Level Supervisor threshold. Reserve agents are then dropped off a skill only when both of the following conditions are met:

- The EWT for the skill drops below both administered thresholds.
- The head call's time in queue no longer exceeds the Service Level Supervisor threshold.

Least Occupied Agent (LOA)

Distributes the calls evenly across all available agents to balance the workload among the agents with fewer skills and Agents with several skills. LOA solves the problem of agents who were bombarded with calls after logging into a skill at the start of a shift, while the agents who are already logged-in have maintained their current incoming call level.

Logged-In Advocate Agent Counting

Counts agents toward the Advocate agent limit if Service Objective, Percent Allocation, or a Reserved Skill is assigned to the agent's login ID, or if one of the agent's skills is assigned Least Occupied Agent or Service Level Supervisor.

Voice Response Integration (VRI)

Integrates Call Vectoring with the capabilities of voice response units such as the Lucent Technologies CONVERSANT Voice Information System. You can also integrate a voice response unit with ACD. All this provides a variety of advantages. For example, while a call is queued, a caller can listen to product information via an audiotext application or can complete an interactive voice-response transaction. It may be possible to resolve the caller's questions while the call is queued, which helps reduce queuing time for other callers during peak times.

Call Charge Information

DEFINITY ECS provides two ways to know the approximate charge for outgoing calls:

- Advice of Charge — For ISDN trunks

  Advice of Charge collects charge information from the public network for each outgoing call. Charge advice is a number representing the cost of a call; it is recorded as either a charging or currency unit.

- Periodic Pulse Metering — For non-ISDN trunks

  Periodic Pulse Metering accumulates pulses transmitted from the public network at periodic intervals during an outgoing call. At the end of the call, the number of pulses collected is the basis for determining charges.

Call-charge information helps you to account for the cost of outgoing calls without waiting for the next bill from your network provider. This is especially important in countries where telephone bills are not itemized. You can also use this information to let employees know the cost of their phone calls, encouraging them to save money on toll calls.

ASAI (Adjunct Switch Application Interface)

CallVisor Adjunct Switch Application Interface (ASAI)

Links DEFINITY ECS and adjunct applications. The interface allows adjunct applications to access DEFINITY ECS features and supply routing information to the system. CallVisor ASAI improves ACD agents' call handling efficiency by allowing an adjunct to monitor, initiate, control, and terminate calls on the switch. The CallVisor ASAI interface may be used for Inbound Call Management (ICM), Outbound Call Management (OCM), and office automation/messaging applications. It uses two transport types: ISDN-BRI transport (CallVisor ASAI-BRI) and LAN Gateway Transmission Control Protocol/Internet Protocol transport (DEFINITY LAN Gateway TCP/IP). CallVisor ASAI messages and procedures are based on the ITU-T Q.932 international standard for supplementary services.

Direct Agent Announcement

Direct Agent Announcement (DAA) enhances Direct Agent Calling capabilities for CallVisor Adjunct-Switch Application Interface (ASAI) and Expert Agent Selection (EAS). It plays an announcement to Direct Agent callers waiting in a queue.

Flexible Billing

Allows DEFINITY ECS or an adjunct to communicate with the public network using ISDN PRI messages to change the billing rate for an incoming 900-type call. Rate-change requests to specify a new billing rate can be made anytime after a call is answered and before it disconnects.

Flexible Billing is available in the U.S. for use with AT&T MultiQuest® 900 Vari-A-Bill™ Service. Flexible billing requires a CallVisor Adjunct-Switch Application Interface and other application software.

ASAI Pending Work Mode Change

Allows ASAI applications to change the current work mode of an agent while that agent is busy on a call. The change is a pending change that will take effect as soon as all the current calls are cleared.

ASAI Trunk Group Identification

Provides ASAI applications with the capability to obtain Trunk Group information even when the Calling Party Number (CPN) is known. ASAI will provide the Trunk Group information in the Event Reports for both inbound and outbound calls. If the ANI is known, the Event Reports will contain the Trunk Group information and the CPN.

CTI (Computer Telephony Integration)

CentreVu Computer Telephony on MAPD-NT

Lucent developed a telephony server that provides a platform against which the Computer Telephony applications may run against to control a DEFINITY ECS switch. The Telephony applications supported include CallVisor PC, TSAPI and JTAPI. Provides one co-resident product for industry standard hardware and Operating System to be able to run in the ProLogix or DEFINITY ECS cabinet. Runs on Windows NT.

CentreVu IP Agent

This is a PC-based IP application that allows agents to use their PCs as phones. In addition to the traditional functionality of a standard DEFINITY phone (transfer, hold, conference, and so forth), IP Agent offers directory services, screen pops, call history, and agent mode history.

ATM

ATM Trunking

Provides Call centers with an alternative to T1/E1 facilities for various private networking configurations

ATM WAN PNC

Enables you to merge together separate, geographically-dispersed DEFINITY ACDs into a single DEFINITY ACD. This can be done by interconnecting port networks over an ATM WAN.

8 —Private Networking Features

Private Network Access

Allows calls to other systems in a private network. These calls do not use the public network. They are routed over your dedicated facilities.

Node Number Routing

Allows you to specify the route pattern associated with each node in a private network. It is a required capability for Extension Number Portability and is used in conjunction with Automatic Route Selection, AAR and ARS Partitioning, Private Networking, and Uniform Dial Plan. Uniform Dial Plan extensions can be routed to a specified node using its associated pattern. Node Number Routing allows a Uniform Dial Plan route pattern based on node numbers or on location codes. On the AAR and ARS Digit Analysis Tables, you also can specify a Node Number instead of a Route Pattern.

ATM-Port Network Connectivity (ATM-PNC)

ATM Port Network Connectivity (ATM-PNC) provides an alternative to either the direct connect or Center Stage Switch configurations for connecting the Processor Port Network (PPN) to one or more Expansion Port Networks (EPNs). ATM PNC replaces the Center Stage Switch in a DEFINITY R8r network with an Asynchronous Transfer Mode (ATM) switch. ATM PNC is available with all three DEFINITY ECS reliability options — standard, high, and critical.

ATM PNC integrates delivery of voice, video, and data via ATM over a common large bandwidth LAN, providing reduced infrastructure cost and improved network manageability. ATM PNC uses standards-based open interfaces that can be provisioned with either new or existing DEFINITY ECS systems and is ATM-ready for future expansion.

ATM-Circuit Emulation Service (ATM-CES)

ATM-CES (Circuit-Emulation Service) lets the DEFINITY ECS emulate an ISDN-PRI trunk on an ATM facility. These virtual trunks can serve as integrated access, tandem, or tie trunks. ATM-CES trunk emulation maximizes port network capacities by consolidating trunking. For example, the CES interface can define up to ten virtual circuits for tie-line connectivity, consolidating onto one circuit card network connectivity that usually requires multiple circuit packs.

ATM Wide Area Network PNC (ATM WAN-PNC)

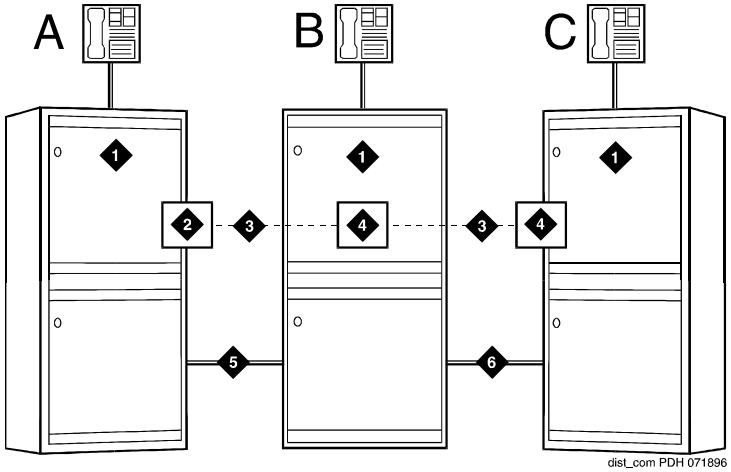
ATM Wide Area Network (ATM WAN) extends the Port Network Connectivity (PNC) beyond a single ATM switch. This allows you to utilize either a private ATM network, public WAN, or a combination of both. Several networked ATM devices can be used as effectively as a single ATM switch for inter-port network connectivity. ATM WAN is supported by the "Multiple Location" feature, where port network cabinets can be administered as separate locations; however, it is not required. You can use multiple ATM switches without multiple locations administered.

Distributed Communications System (DCS)

Distributed Communications System (DCS) allows you to configure 2 or more DEFINITY switches as if they were a single, large DEFINITY ECS. DCS provides attendant and voice-terminal features between these switch locations. DCS simplifies dialing procedures and allows transparent use of some of the DEFINITY ECS features. (Feature transparency means that features are available to all users on DCS regardless of the switch location.)

See also, “Centralized Attendant Service” on page 47.

See also, “Inter-PBX Attendant Calls” on page 48.



1 DEFINITY ECS: DCS node

2 Processor Interface

3 Data links (TCP/IP protocol)

4 Packet Gateway

5 Tie Trunk: DS1

6 Tie Trunk: ISDN-Primary Rate Interface switched network or private line

Figure 14. DEFINITY ECS as Part of a DCS

Uniform Dial Plan (UDP)

A unique four- or five-digit number assigned to each station on the network. Uniform numbering gives each station a unique number (location code plus extension) that can be used at any location in the Electronic Tandem Network to access that station. DEFINITY ECS enhances the standard UDP with the unrestricted 5-digit Uniform Dial Plan, which allows up to five digits to be parsed for call routing.

Attendant Features That Work Over the Network

- Attendant Control of Trunk Group Access
- Attendant Direct Trunk Group Selection
- Phone Displays
- DCS Busy Verification of Terminals and Trunks
- DCS Trunk Group Busy/Warning Indication
- Attendant Vectoring

Phone Features That Work Over the Network

- DCS Alphanumeric Display for Terminals
- DCS Attendant Display
- DCS Automatic Callback
- DCS Coverage Callback
- DCS Call Coverage
- DCS Distinctive Ringing
- DCS Leave Word Calling
- DCS Multiappearance Conference/Transfer
- Attendant Crisis Alert
- Emergency Access to the Attendant
- DCS Call Waiting

System Management Features That Work Over the Network

- Calling/Connected Party Number (CPN) Restriction
- Automatic Circuit Assurance

DCS Over ISDN-PRI D-channel

Enhances DCS by allowing access to the public network for DCS connections between DCS switch nodes. With this feature (also known as DCS+), DCS features are no longer restricted to private facilities. The ISDN-PRI B-channel is used for voice communications, and the ISDN-PRI D-channel is used to transport DCS control information.

QSIG

QSIG provides compliance to the International Organization for Standardization (ISO) ISDN-PRI private-networking specifications. QSIG is defined by ISO as the worldwide standard for private networks. QSIG features are supported on BRI trunks.

QSIG is the generic name for a family of signaling protocols. The Q-reference point or interface is the logical point where signaling is passed between 2 peer entities in a private network. QSIG signaling can provide feature transparency in a single-vendor or multi-vendor environment.

QSIG provides call-related Supplementary Services. These are services that go beyond voice or data connectivity and number transport and display. Examples of Supplementary Services include Name Identification, Call Forwarding (Diversion), and Call Transfer.

Call Completion

Call Completion utilizes the QSIG Platform enhancement Call Independent Signaling Connections and is functionally equivalent to the Distributed Communications System (DCS) feature: AutoCallback.

Call Independent Signaling Connections (CISC)

Call Independent Signaling Connections (CISC) are used to pass QSIG Supplementary Service information that is independent of an active call between two QSIG compliant nodes. Implementation is based on the ISO standard for CISC. It is possible to determine the status of a QSIG TSC by using the "status Signaling group" command on the SAT.

Manufacturer-Specific Information (MSI)

QSIG handles non-standardized information that is specific to a particular PBX or network. This information is known as Manufacturer Specific Information (MSI). A manufacturer can define manufacturer-specific supplementary services operations after it has:

- Applied to a sponsoring and issuing organization (ECMA or European Computer Manufacturers Association in this case)
- Been assigned an organization identifier. This organization identifier is used as the root of the manufacturer-specific service-operation value.

All MSI operation values should be unique to that manufacturer.

Manufacturer-specific supplementary services can be created using specific operations encoded with the manufacturer's identifier. DEFINITY ECS supports non-QSIG applications that transport information across QSIG networks in MSI. Applications now have the same functionality over QSIG networks that they have over non-QSIG networks. Applications that use MSI include Centralized Attendant Service, Transfer to Audix, Best Service Routing, and QSIG VALU.

Path Replacement

With this feature, a call's connections between switches in a private network can be replaced with new connections while the call is active. This feature is invoked when a call is transferred and improvements may be made in costs. For example, after a call is transferred, the two parties on the transferred call can be connected directly and the unnecessary trunks are dropped off the call. The routing administered at the endpoints may allow for a more cost-effective connection.

Call Forwarding (Diversion)

QSIG Call Forwarding (Diversion) is based on the DEFINITY ECS Call Forwarding feature. It extends the feature transparency aspects of Call Forwarding over a QSIG trunk:

- If QSIG Call Forwarding is activated, all calls are diverted immediately.
- If QSIG Call Forwarding with Busy/Don't Answer is activated and a station is busy, a call is diverted immediately.

- If QSIG Call Forwarding with Busy/Don't Answer is activated and a station is idle but the call is not answered, a call is diverted after a specified number of rings.

These features are activated either by dialing a Feature Access Code (FAC) or by pressing a button. See Call Forwarding for detailed descriptions of how to use these features.

Call Transfer

QSIG Call Transfer differs from the standard DEFINITY ECS Transfer feature in that additional call information is available for the connected parties after the transfer completes. However, the information is only sent for QSIG trunks. If one call is local to the transferring switch, that user receives the name of the party at the far end.

Name and Number Identification

Allows a switch to send and receive the calling number, calling name, connected number, and connected name. Additional parameters that control the display of the connected name and number are administered on the Feature-Related System-Parameters form. QSIG Name and Number Identification displays up to 15 characters for the calling and connected name and up to 15 digits for the calling and connected number across ISDN-PRI interfaces.

Called NAME ID

The QSIG Called Name feature presents the called party's name on the calling party's display while the call is ringing. It then reverts to "connected name" when answered.

Call Offer

This feature, on request from the calling-user (or on that user's behalf), enables a call to:

- Be offered to a busy called-user
- Wait for a busy called-user to accept the call when the necessary resources have become available

QSIG Centralized Attendant Service (CAS)

Provides you with the capability to have all your Attendants in one location, serving users in multiple locations. QSIG CAS does not utilize separate Release Link Trunks (RLT). This feature will not restrict calls from going out over non-QSIG trunks; however, the full functionality of the QSIG CAS will not be available.

RLT Emulation via a PRI

ISDN QSIG trunks will route calls from the branch PBX to the main PBX. You no longer have to specify a dedicated RLT network. The QSIG Path replacement takes care of the trunk optimization. You have the flexibility to route calls to the main PBX.

Attendant Display of Class of Restriction (COR)

While on a call, the Attendant can press a "COR display" button to see the Class of Restriction of the user. The Attendant will not block the transfer of the restricted line to the user. This feature is used for informational purposes only.

Attendant Return Call

If a call that is transferred by the Attendant goes unanswered for a specified period of time, the call is returned to the Attendant. Preferably the call will transfer back to the Attendant who transferred the call.

Priority Queue

QSIG MSI will pass more information to the main PBX. This information enables calls coming in from a QSIG CAS branch to be placed in the appropriate place in the queue, as if the call originated on the main PBX.

QSIG VALU

Distinctive Alerting

Provides distinctive ringing, internal and external, to the remote called party when the call is routed over the QSIG network.

Call Coverage

Provides similar call coverage as DCS Call Coverage and Call Coverage Remote Off Net or C-CRON. Call will come back if covered over QSIG. The functionality will only be complete when all the switches are DEFINITY ECS and using QSIG VALU. The Covered-to party can still receive Distinct Alerting.

Call Coverage and CAS

When a trunk has both CAS and VALU Call Coverage activated, the coverage display information is provided on calls that cover from a branch PBX to the main PBX. Path replacement will be attempted after coverage.

QSIG Voice Mail Capabilities

QSIG voice mail allows users on a remote switch to cover to a voice mail system on another switch, provided the voice mail system uses a QSIG interface. Users receive a message-waiting indication (message lamp or stutter dial tone) and have access to most functions of the voice mail system. When the DEFINITY ECS and a voice mail system are connected via QSIG, users can transfer directly into a mailbox on the voice mail system. Transfer to voice mail is activated by a feature access code and requires QSIG MSI

9 —Adjunct Systems

Voice Messaging and Response

DEFINITY AUDIX

While many voice messaging systems require separate equipment and connections, the DEFINITY AUDIX System easily installs directly into your DEFINITY ECS cabinet to support advanced voice messaging capabilities without the need for an adjunct processor.

Each DEFINITY AUDIX system supports up to 2000 mailboxes and stores up to 100 hours of recorded messages. It can be configured with 2 to16 ports (in two-port increments).

Whenever you call the DEFINITY AUDIX system, you interact with it by entering commands through your telephone's touch-tone keypad. You simply specify the desired activity, and follow the voice prompts for the desired task.

Special voice-processing features include Voice Mail, Call Answering, Outcalling, Multi-Level Automated Attendant, and Bulletin Board. The following is a summary of DEFINITY AUDIX capabilities:

- *Shared Extensions* provide personal mailboxes for each person sharing a phone
- *Multiple Personal Greetings* allows you to prepare a pool of up to nine personal greetings to save time and provide more personal customer service. Separate messages can indicate you are on the phone, away from the desk, on vacation, etc. You can assign different messages to internal, external, or after-hours calls
- *Priority Messaging* places important messages ahead of others. Internal and External callers can mark the message as priority.
- *Outcalling* automatically dials a prearranged phone number or pager when you have messages in your voice mailbox
- *Priority Outcalling* automatically dials a prearranged phone number or pager when you have *priority* messages in your voice mailbox
- *Broadcasting* allows you to send a single message to multiple recipients or to all users on the system

- *System Broadcast* allows you to send broadcast messages as regular voice messages, or as messages that recipients hear as they log in
- *AUDIX Directory,* allows you to look up the extension number of any other user by entering their name on the telephone keypad
- *Personal Directory* allows you to create a list of nicknames for quick access to telephone numbers
- *Call Answering for Nonresident Subscribers* provides voice mailboxes for users who do not have an extension number on the DEFINITY ECS
- *Full Mailbox Answer Mode* informs callers whenever messages cannot be left because there is no room in a subscriber's mailbox
- *Name Record by Subscriber* lets you record your own name on the system
- *Automatic Message Scan* can play all new messages in part or in their entirety without requiring you to press additional buttons*, which is particularly useful when you are getting messages from your mobile phone*
- *Sending Restrictions by Community* enables you to limit the communities of callers who can communicate via AUDIX Voice Messaging
- *Group Lists* allows you to create mailing lists of up to 250 people to use for broadcasting messages
- *Message Forwarding* allows you to forward messages with or without attached comments
- *Name Addressing* allows you to address messages by name if you don't know the extension
- *Private Messaging* is a special coding feature that prevents recipients from forwarding messages
- *Leave Word Calling* allows you to press a button on your telephone in order to leave a standard *call me* message on any extension
- *Online Help* provides you with instant access to voiced instructions at any time when you are using the system.
- *Multiple Language Support* allows you to install up to nine languages on the system, from a superset of 30 available languages.
- *Enhanced Message Handling* gives you the flexibility for handling messages. Two of these features are Optional Advance/Rewind, which lets you advance through and rewind individual messages, and Undelete Messages, which lets you retrieve any messages that you may have accidentally deleted.

INTUITY AUDIX

INTUITY Messaging Solutions essentially offers the same user features as the DEFINITY AUDIX System, plus the following features:

- *Fax Messaging* allows you to handle faxes as easily as you handle voice mail. You can send, receive, store, scan, delete, skip, or forward faxes. This feature is fully integrated with voice messaging, so you can attach faxes to voice messages, for example. You can also create special mailboxes for each of your fax machines. These mailboxes accept fax telephone calls when the fax machine is busy and then deliver the fax to the fax machine when the fax machine is available
- *Turn off AUDIX Call Answering* allows you to turn off Call Answering in order to conserve system resources. You can create a message that tells callers they cannot leave a message, giving them another number to call, for example
- *Pre-Addressing* allows you to address a message before recording it
- *Integrated Messaging* allows you access and manage incoming voice, fax, and e-mail messages and file attachments from your personal computer or your telephone. A voice message will thus appear in your e-mail mailbox, for example, and vice versa. You can also set options to have just the message headers appear in the alternate mailbox. You can also create a voice or fax message by telephone and send it to an e-mail recipient
- *Text-to-Speech* allows you listen to a voice rendering of text messages sent from a supported e-mail system and/or INTUITY Message Manager
- *Print Text* allows you to print messages sent from a supported e-mail system and/or INTUITY Message Manager
- *Enhanced Addressing* allows you to send a message to up to 1500 recipients
- *Transfer Restrictions* allow you to control toll fraud by restricting transfers going through the voice messaging system
- *Internet Messaging* allows you to exchange messages (voice and text) with any e-mail address via the World Wide Web.
- *Lucent Voice Director* allows you to address messages via spoken name, in addition to using touchtones to enter extensions or names. It also supports transferring to AUDIX subscribers, including those in other locations, by speaking a name.

Mode Code Interface

DEFINITY ECS supports an analog Mode Code interface for communications with INTUITY AUDIX and other voice mail systems using the same interface. This interface employs DTMF tones, line signals, and feature access codes, and allows INTUITY AUDIX to exchange data with the DEFINITY ECS without using a data link. Other adjunct vendors can engineer their products to use this interface.

Dual DCP I-Channels

Support the use of dual DCP I-channels for AUDIX networking. In this case, networking refers to the ability to send data files between AUDIX systems, not to communications with the switch.

INTUITY Lodging

Lucent Technologies INTUITY Lodging is a messaging system designed especially for lodging establishments such as hotels or other lodging providers such as hospitals or colleges. The system supplies guests with electronic mailboxes that store voice or fax messages. INTUITY Lodging serves as a private answering machine for each extension.

Hotel guests can leave messages for each other without going through the Attendant. For incoming calls, an Attendant transfers the call to the appropriate room. If the guest does not answer the call or if the line is busy, the call is automatically transferred to the guest's voice mailbox, where the caller can leave a voice message.

A message-waiting indicator on the guest's phone notifies the guest that the voice mailbox contains messages. Guests are assigned a password for accessing messages remotely. They can retrieve and save messages from any telephone, on or off premises.

INTUITY Conversant

The INTUITY CONVERSANT Voice Information System is an interactive voice-response system that automates phone-call transactions from simple tasks ,like routing to the right department, to complex tasks, such as registering college students or providing bank balances. It communicates with customers in natural-sounding, digitally recorded speech. And it performs — 24 hours a day and without the services of an operator.

The system can handle single or multiple voice-response applications simultaneously, and can serve up to 48 callers at once. It can operate by itself to dispense information or collect data, or it can work with a host computer to access a large database such as bank account records. With its speech-recognition capability, even rotary telephone users can

have access to sophisticated phone-based services. Advanced telephone features provide intelligent call-transfer capabilities and allow you to use the system in your existing telephone environment.

INTUITY Call Accounting System

If you are using any of the INTUITY voice messaging products, the INTUITY Call Accounting System is probably the best call-accounting solution for you. The system works exclusively with INTUITY products, which reside on MAP/40 or MAP/100 computers. Offering many of same features as the Call Accounting System for Windows (described in the previous section), the system also serves to help integrate your INTUITY applications.

Centralized Voice Mail via Mode Code Integration

The Centralized Voice Mail feature eliminates the need for a voice mail system at each of the sites in a network. It does so by allowing a DEFINITY ECS network to use a single DEFINITY AUDIX or INTUITY AUDIX or Octel 100 Voice Messaging System as a centralized voice mail system that serves the whole network. The INTUITY AUDIX or Octel 100 system can also serve as a centralized voice mail system within a hybrid network of DEFINITY and Merlin Legend/Magix switches. The DEFINITY and Merlin pass information to each other via mode codes.

Other Supported Systems

AUDIX Voice Power (AVP)

The AUDIX Voice Power (AVP) System from Lucent Technologies is a cost-effective addition to your INTUITY CONVERSANT® or CONVERSANT INTRO system, providing 24-hour-a-day voice messaging communications to individuals or departments.

AUDIX Voice Power Lodging (AUDIX VPL)

AUDIX Voice Power Lodging (AUDIX VPL) is a voice mail system designed especially for lodging establishments such as hotels. It supplies guests with electronic mailboxes that store voice messages. AUDIX VPL is like having private answering machines that take messages for guests when they are unavailable.

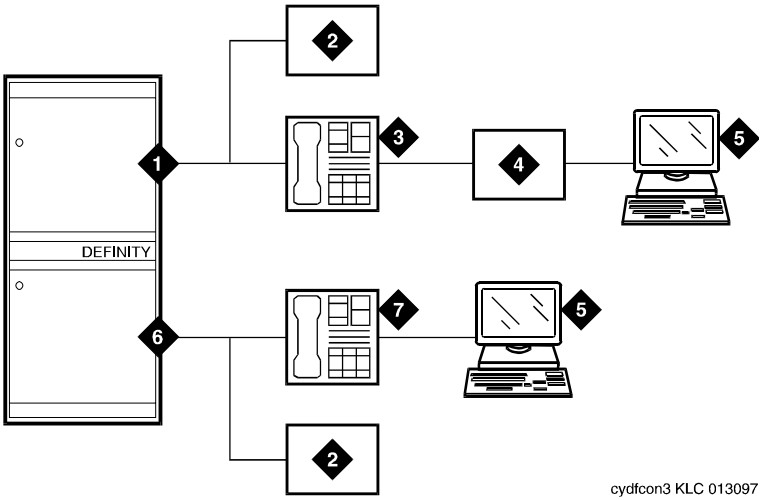
Call Center (See Call Center Features)

PassageWay

Lucent Technologies PassageWay products bring the telephone and the personal computer together into an integrated voice and data workstation that can greatly enhance communications.

PassageWay Direct Connection links your company's desktop personal computers with an easy-to-use Microsoft Windows interface to give you greater business communications capabilities than either the telephone or the personal computer offer alone.

PassageWay Direct Connection is well-suited for those users who are constantly conducting business using both a Windows-based personal computer and a Lucent Technologies telephone and want to boost their efficiency.



1 2- or 4-wire DCP Port
2 Auxiliary Power
3 DCP Telephone
4 Passageway
5 Personal Computer
6 2-wire DCP Port
7 8411 DCP Telephone w/ PassageWay

Figure 15. PassageWay Direct Connect Configurations

Wireless

The DEFINITY Wireless Business System relies on the DEFINITY ECS system to manage mobility. It uses Personal Wireless Telecommunications technology, which is a leading protocol in the United States. This standard, which has the primary advantage of permitting up to 12 simultaneous conversations per base station, defines the radio interface between the portable telephones and the base stations in the system.

The DEFINITY Wireless Business System is fully integrated with the DEFINITY ECS, and offers users full access to the DEFINITY ECS features. The system has the following maximum capacities:

- 260 wireless telephones
- 60 base stations
- 7,000 to 40,000 calls per busy hour (depending on DEFINITY ECS configuration)
- 4 million square foot (1.2 million square meter) coverage area

Forum Personal Communications Manager

Forum's System Manager provides superior system administration capabilities. The Forum Personal Communications Manager can accommodate even the largest businesses. It has the following maximum capacities:

- 500 wireless telephones
- 126 base stations
- 6 PRI interfaces
- 1 Sun workstation
- 4 million square foot (1.2 million square meter) coverage area

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